

*This specification describes the situation of the Proximus network and services. It will be subject to modifications for corrections or when the network or the services will be modified. Please take into account that modifications can appear at any moment. Therefore, the reader is requested to check regularly with the most recent list of available specifications that the document in one's possession is the latest version.*

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**IMS CORPORATE VoIP  
SIP SIGNALLING**

**Business Trunking with IMS services**

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## 0. Document History

Every update of this document results in a complete new version with new version number and release date.

Version	Date	Main or important changes since previous version
1.0	February 14, 2011	First version
1.1	February 16, 2011	Correction/addition regarding accepted number formats
1.2	March 02, 2011	Addition emphasizing the importance to use a domain in the Request-URI (rather than an IP address)
1.3	May 16, 2011	Correction to the cover sheet
1.4	June 29, 2011	References added for support of multipart message bodies Updates regarding P-Asserted-ID handling
1.5	July 04, 2011	Redundancy procedures added
1.6	September 05, 2011	P-Asserted-ID removed from examples because IMS does not expect to receive this header from an IP-PBX
1.7	December 20, 2011	Domain in Request-URI replaced by IP address of IP-PBX for calls towards IP-PBX
1.8	March 26, 2012	Addition regarding proprietary SIP headers
1.9	October 9, 2012	Addition of a warning concerning the use of the Retry-After header in 503 response
1.10	Januari 13, 2016	Changed "Belgacom" in "Proximus"
1.11	September 05, 2016	Made support of both methods for DTMF transport mandatory
1.12	November 24, 2016	Update due to name change "Bizz IP telephony multi" into "Enterprise voice multi"
1.13	June 28, 2018	Clarification concerning the Max-Forwards header The use of Diversion header added.
2.0	December 18, 2018	Example flows updated
2.1	April 17, 2019	Moved chapter on emergency to PXM IMS Corporate VOIP - UNI specification – General Corrected typo errors Included International Business Trunking information
2.2	May 25, 2019	Document name changed from BGC to PXM
2.3	June 06, 2019	Small corrections based on comments from the IP-PBX team
2.4	September 19, 2019	Update of SW versions in Scope
2.5	November 20, 2020	Update concerning the use of the Privacy header

2.6	February 06, 2023	Example flows correction
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Table 1: document history

## 1. Scope

This document defines the SIP signalling over the VoIP interface between the Proximus IMS Network and large IP-P(A)BXs, connected as SIP Business Trunk or as SIP International Business Trunk. The specifications listed in this document are not exhaustive but have to be interpreted as “minimal requirements for compliance to the Proximus IMS Corporate VoIP services”.

The specifications are applicable for the following IMS equipment and software packages:

Nokia (Alcatel-Lucent) ISC – software package Release 13  
Oracle SBC 4600 – Software Version SCZ8.1.0 MR-1 patch 12  
Broadsoft application server – BroadWorks R22

This document is part of a set of documents describing the UNI interface of the Proximus IMS Network, for IP-P(A)BXs. Other documents in this set are:

PXM IMS Corporate VoIP – UNI specification – General [1]  
PXM IMS Corporate VoIP – UNI specification – SIP signaling – Wireless Office Extended [2]  
PXM IMS Corporate VoIP – UNI specification – SIP signaling – Enterprise Voice Multi [3]  
PXM IMS Corporate VoIP – UNI specification – Testing  
PXM IMS VoIP – UNI specification – Fax over IP [36]  
PXM IMS Corporate VOIP - UNI specification - Address templates [40]

## 2. References

Whenever a date of edition is mentioned, the document with this date should be consulted. If no date is present, the latest version of this document should be consulted.

### 2.1. Normative references

1	Proximus	PXM IMS Corporate VoIP – UNI specification – General
2	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Wireless Office Extended
3	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Enterprise Voice Multi
4	3GPP TS 24.229	IP Multimedia call control protocol based on SIP and SDP, Stage 3 Release 9
5	3GPP TS 23.167	IP Multimedia Subsystem (IMS) emergency sessions, Release 9
6	SIP forum	The SIP connect 1.1 technical recommendation (draft)
7	ETSI TS 182 025	Business Trunking; architecture and functional description v2.1.1
8	ITU-T E.164	The international telecommunication numbering plan
9	IETF RFC 3261	SIP: Session Initiation Protocol
10	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
11	IETF RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
12	IETF RFC 3265	Session Initiation Protocol (SIP)-Specific Event Notification
13	IETF RFC 4566	SDP: Session Description Protocol
14	IETF RFC 2976	The SIP INFO Method
15	IETF RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method
16	IETF RFC 3323	A Privacy Mechanism for the Session Initiation Protocol (SIP)
17	IETF RFC 3325	Private Extensions to SIP for Asserted Identity within Trusted Networks
18	IETF RFC 3326	The Reason Header Field for the Session Initiation Protocol (SIP)
19	IETF RFC 3455	Private header extensions for SIP for 3GPP
20	IETF RFC 3515	The Session Initiation Protocol (SIP) REFER Method
21	IETF RFC 3891	The Session Initiation Protocol (SIP) “Replaces” Header
22	IETF RFC 3892	The Session Initiation Protocol (SIP) Referred-By Mechanism
23	IETF RFC 3960	Early Media and Ringing Tone Generation in SIP
24	IETF RFC 4028	Session Timers in the Session Initiation Protocol (SIP)
25	IETF RFC 2617	HTTP Authentication: Basic and Digest Access Authentication
26	IETF RFC 1321	The MD5 Message-Digest Algorithm
27	IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications
28	IETF RFC 3551	RTP Profile for Audio and Video Conferences with Minimal Control
29	IETF RFC 4733	RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
30	IETF RFC 4734	Definition of Events for Modem, FAX and Text Telephony signals
31	IETF RFC 5806	Diversion indication in SIP
32	IETF RFC 6044	Mapping and Interworking of Diversion Information between Diversion and History-Info Headers in the Session Initiation Protocol (SIP)
33	IETF RFC 4244	An Extension to SIP for Request History Information
34	IETF RFC 3966	The Tel URI for Telephone Numbers
35	IETF draft	Applying Loose Routing to Session Initiation Protocol (SIP) User Agents (UA)” (expired)
36	Proximus	PXM IMS VoIP – UNI specification – Fax over IP
37	ETSI TS 183 011	Anonymous communication rejection (ACR) and communication barring (CB)
38	IETF RFC 5621	Message body handling in SIP
39	IETF RFC 5876	Updates to Asserted Identity in the Session Initiation Protocol (SIP)
40	Proximus	PXM IMS Corporate VOIP - UNI specification - Address templates

Table 2: normative references



## 2.2. Informative references

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Table 3: informative references

## 3. Symbols, Definitions and Abbreviations

### 3.1. Symbols

For the purpose of the present document, the following symbols apply:

None.

### 3.2. Definitions

For the purpose of the present document, the following definitions apply:

**IP-P(A)BX:** The IP P(A)BX constitutes an Enterprise's collection of network elements that provides packetized voice call origination and termination services using the Session Initiation Protocol (SIP) and the Session Description Protocol (SDP) for signalling and the Real-time Transport Protocol (RTP) for media traffic.

**pbxPUID:** The public user identity referring to the IP P(A)BX as a whole. The pbxPUID will/can be used as host part of the SIP URI used by the IP-P(A)BX.

**PBXName:** The same as pbxPUID.

**Dialled-SubB-dn:** destination number as dialled by the originating user. The format can be:

- +<CC><NSN>
- 0 <NSN>
- 00<CC><NSN>
- <1AB> (emergency numbers)
- <1ABC> (short code services)

**Norm-SubB-dn:** destination E.164 number in international format (i.e. +<Country Code><Area Code><DN> ex: +32227970231).

**Norm-SubA-dn:** originating E.164 number in international format (i.e. +<Country Code><Area Code><DN> ex: +32227970231).

**Dialled-SubC-dn:** forwarded -to number as programmed by the forwarding user. The format can be the same as the Dialled-SubB-dn

**Norm-SubC-dn:** forwarded-to E.164 number in international format (i.e. +<Country Code><Area Code><DN> ex: +32227974563).

**DisplayName:** the name of the user.

**EnterpriseDomain:** the public domain name used by the enterprise. Currently the default domain name is ims.belgacom.be. The possibility for the IP-P(A)BX to use, in the future, as public domain name it's own domain name e.g. mycompany.com is under study.

Note: EnterpriseDomain may also be an IPaddress instead of a domain name

**IP-addr-PBXName:** the IP address of the PBX with name PBXName.

**IP-addr-IMS:** the IP address of the Proximus IMS network access point (i.e. the SBC).

**1AB:** called emergency service e.g. 100,112, etc.

**1ABC:** called short code service

### 3.3. Abbreviations

See §2.2 of “PXM IMS Corporate VoIP – UNI specification – General” [1]

Additionally for the purpose of the present document, the following abbreviations apply:

CC	Country Code
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Presentation Restriction
CFU	Call Forwarding Unconditional
CFB	Call Forwarding on Busy
CFNR	Call Forwarding on No Reply
DN	Directory Number
DTMF	Dual Tone Multi-Frequency
iDN	Individual directory number i.e. a particular DN within the range of the IP-P(A)BX
OCB	Outgoing Call Barring
NSN	National Significant Number
SBC	Session Border Controller

## 4. General

### 4.1. Structure of the document

The general structure of this document mainly covers the following aspects:

- SIP profile: this chapter will contain the list of RFCs and standards to which the Proximus IMS network complies, relevant for IP-P(A)BX interconnection.
- SIP behaviour: besides the protocol support also the expected behaviour is important. E.g. how do supplementary services behave? Main topics to be covered in this chapter:
  - Registration/authentication
  - Basic Call
  - Supplementary services
  - Call admission control
  - CLI screening
  - Fax support
  - Emergency calls
  - DTMF
  - Redundancy procedures

### 4.2. Reference point

See §5.3.1 and §5.3.3 of [1]: “PXM IMS Corporate VoIP – UNI specification – General”.

## 5.SIP Profile

### 5.1. Introduction

The following clauses list the SIP related 3GPP and ETSI standards and IETF RFCs to which the IP-P(A)BX shall be compliant.

### 5.2. 3GPP standards

The Proximus IMS network implements and supports SIP protocol in accordance with:

- TS 24.229 IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP), stage 3 Release 7 [4]
- TS 23.167 IP multimedia subsystem (IMS) emergency sessions, Release 7 [5]  
Although this is a registration configuration, emergency registrations are not applicable.

### 5.3. ETSI TISPA standards

The Proximus IMS network implements and supports SIP protocol in accordance with:

- TS 182 025 Business Trunking; architecture and functional description”, v2.1.1 (2008-09) [7]

### 5.4. IETF RFCs

The Proximus IMS network implements and supports SIP/SDP protocol in accordance with:

- RFC 3261 SIP: session initiation protocol [9]
- RFC 3262 Reliability of provisional responses in SIP [10]
- RFC 3264 an offer/answer model with SDP [11]
- RFC 3265 SIP-specific event notification [12]  
Only the NOTIFY method is supported, the SUBSCRIBE method and the 489 “bad event” response are not supported.  
Because the NOTIFY method is a rather generic method usable in many different functions (e.g. message waiting indication) it can not be guaranteed that the behaviour will be correct for every possible functionality using the NOTIFY method.
- RFC 4566 SDP: session description protocol [13]
- RFC 2976 SIP INFO method [14]  
Because the INFO method is a rather generic method usable in many different functions it can not be guaranteed that the behaviour will be correct for every possible functionality using the INFO method. For example DTMF transport in the INFO method

doesn't work properly. Therefore it is mandatory to support the mechanisms for DTMF transport listed in § 6.9

- RFC 3311 SIP UPDATE Method [15]
- RFC 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP) [16]
- RFC 3325 Private Extensions to SIP for Asserted Identity within Trusted Networks [17]
- RFC 3326 SIP reason header [18]
- RFC 3455 3GPP P-headers [19]
- RFC 3515 SIP REFER Method [20]
- RFC 3891 The Session Initiation Protocol (SIP) "Replaces" Header [21]
- RFC 3892 The Session Initiation Protocol (SIP) Referred-By Mechanism [22]
- RFC 3960 Early Media and Ringing Tone Generation in SIP [23]
- RFC 4028 Session Timers in the SIP [24]
- RFC 2617 HTTP Authentication: Basic and Digest Access Authentication [25]
- RFC 1321 The MD5 Message-Digest Algorithm [26]
- RFC 3550 RTP: A Transport Protocol for Real-Time Applications [27]
- RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control [28]
- RFC 4733 RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals [29]

This RFC is the successor of RFC 2833. Please refer to the abstract of RFC 4733 for indications about backwards compatibility.

- RFC 4734 Definition of Events for Modem, FAX and Text Telephony signals [30]
  - RFC 5806 Diversion indication in SIP [31]
  - RFC 6044 Mapping and Interworking of Diversion Information between Diversion and History-Info Headers in the Session Initiation Protocol (SIP) [32]
  - RFC 4244 An Extension to SIP for Request History Information [33]
  - RFC 3966 The Tel URI for Telephone Numbers [34]
  - Draft Applying Loose Routing to Session Initiation Protocol (SIP) User Agents (UA)" (expired) [35]
  - RFC 5621 Message Body Handling in SIP [38]
- Since more and more SIP applications use multipart message bodies, the correct support and handling of multipart message bodies is very important.
- RFC 5876 Updates to Asserted Identity in the Session Initiation Protocol (SIP) [39]

### 5.4.1. Supported methods

The Proximus NGN network supports the following methods:

- INVITE according to [4] and [9]
- ACK according to [4] and [9]
- BYE according to [4] and [9]
- CANCEL according to [4] and [9]
- REGISTER according to [4] and [9]
- OPTIONS according to [4] and [9]
- PRACK according to [4] and [10]
- NOTIFY according to [4] and [12]
- REFER according to [4] and [20]
- UPDATE according to [4] and [15]
- INFO according to [4] and [14]

### 5.4.2. Supported responses

Provisional responses (1xx)

- 100 Trying according to [4] and [9]
- 180 Ringing according to [4] and [9]
- 181 Call Is Being Forwarded according to [4] and [9]
- 182 Queued according to [4] and [9]
- 183 Session Progress according to [4] and [9]

Successful responses (2xx):

- 200 OK according to [4] and [9]
- 202 Accepted according to [4] and [12]

Redirection responses (3xx):

- 302 Moved Temporarily according to [4] and [9]

Request Failure responses (4xx):

- 400 Bad Request according to [4] and [9]
- 401 Unauthorized according to [4] and [9]
- 402 Payment Required according to [4] and [9]
- 403 Forbidden according to [4] and [9]
- 404 Not Found according to [4] and [9]
- 405 Method Not Allowed according to [4] and [9]
- 406 Not Acceptable according to [4] and [9]
- 407 Proxy Authentication Required according to [4] and [9]
- 408 Request Timeout according to [4] and [9]
- 410 Gone according to [4] and [9]
- 413 Request Entity Too Large according to [4] and [9]
- 414 Request-URI Too Long according to [4] and [9]
- 415 Unsupported Media Type according to [4] and [9]
- 416 Unsupported URI Scheme according to [4] and [9]
- 420 Bad Extension according to [4] and [9]
- 421 Extension Required according to [4] and [9]
- 423 Interval Too Brief according to [4] and [9]
- 480 Temporarily Unavailable according to [4] and [9]
- 481 Call/Transaction Does Not Exist according to [4] and [9]
- 482 Loop Detected according to [4] and [9]
- 483 Too Many Hops according to [4] and [9]
- 484 Address Incomplete according to [4] and [9]
- 485 Ambiguous according to [4] and [9]
- 486 Busy Here according to [4] and [9]
- 487 Request Terminated according to [4] and [9]
- 488 Not Acceptable Here according to [4] and [9]
- 491 Request Pending according to [4] and [9]
- 493 Undecipherable according to [4] and [9]

Server Failure responses (5xx):

- 500 Server Internal Error according to [4] and [9]
- 501 Not Implemented according to [4] and [9]
- 502 Bad Gateway according to [4] and [9]
- 503 Service Unavailable according to [4] and [9]
- 504 Server Time-out according to [4] and [9]
- 505 Version Not Supported according to [4] and [9]
- 513 Message Too Large according to [4] and [9]

Global Failures responses (6xx):

- 600 Busy Everywhere according to [4] and [9]
- 603 Decline according to [4] and [9]
- 604 Does Not Exist Anywhere according to [4] and [9]
- 606 Not Acceptable according to [4] and [9]

### 5.4.3. Supported headers

- Via according to [4] and [9]
- To according to [4] and [9]
- From according to [4] and [9]
- CSeq according to [4] and [9]
- Call-Id according to [4] and [9]
- Contact according to [4] and [9]
- Max-Forwards according to [4] and [9]

As recommended in RFC 3261 [9] clause 20.22 the IP-PBX shall use as initial value of the Max-Forwards header the value 70.

- Route according to [4] and [9]
- Record-Route according to [4] and [9]
- Content-Type according to [4] and [9]
- Content-Disposition according to [4] and [9]
- Content-Length according to [4] and [9]
- MIME-Version according to [4] and [9]
- Supported according to [4] and [9]
- Require according to [4] and [9]
- Expires according to [4] and [9]
- Unsupported according to [4] and [9]
- Allow according to [4] and [9]
- Retry-After according to [4] and [9]
- In-Reply-To according to [4] and [9]
- Authorization according to [4] and [9]
- Proxy-Authenticate according to [4] and [9]
- WWW-Authenticate according to [4] and [9]
- Min-Expires according to [4] and [9]
- Proxy-Authorization according to [4] and [9]
- Alert-Info according to [4] and [9]
- Warning according to [4] and [9]
- Accept according to [4] and [9]
- Call-Info according to [4] and [9]
- RSeq according to [4] and [10]
- RAck according to [4] and [10]
- Allow-Events according to [4] and [12]
- Event according to [4] and [12]
- Subscription-State according to [4] and [12]
- Refer-To according to [4] and [20]
- Privacy according to [4] and [16]
- P-Asserted-Identity according to [4] and [17]
- Reason according to [4] and [18]
- P-Called-Party-ID according to [4] and [19]
- P-Access-Network-Info according to [4] and [19]
- P-Associated-URI according to [4] and [19]
- Replaces according to [4] and [21]
- Referred-By according to [4] and [22]



- Session Expires according to [4] and [24]
- Min-Se according to [4] and [24]
- Diversion according to [31]

#### 5.4.4. Unsupported headers

Proprietary SIP headers (i.e. headers starting with “X-“ or “x-“) are never supported and shall not be sent on the interface.

#### 5.4.5. Supported bodies

Multipart message bodies are supported and used in the Proximus IMS network. Therefore, message body handling according to RFC 5321 [38] is mandatory.

In any case correct support and usage of SIP response 415 “Unsupported Media type” as specified in RFC 3261 [9] is mandatory.

### 5.5. Timer values

The Proximus IMS network supports the timers described in [4] and [9] with the following (default) settings:

- T1 = 500 msec
- T2 = 4 seconds
- T4 = 5 seconds
- Timer D = 32 seconds
- Timer H = 32 seconds

All other timers are derived from these, according to [4] and [9].

### 5.6. Transport protocol

The Proximus IMS network supports SIP over UDP only.

Important remark: In case SIP message length approaches the MTU size, [4] and [9] specify that UDP transport should be replaced by TCP transport. This **shall NOT** be applied by the IP-P(A)BX.

## 6. SIP Behaviour

### 6.1. Introduction

Besides the Standards and RFCs describing the protocol supported, also the expected behaviour is important. E.g. how does registration exactly take place, which information is expected in the “From” header, etc.

Business trunking corresponds to the Peering based business trunking as described in [7].

#### IMPORTANT:

SIP signalling serves the call set-up/teardown of calls/sessions as well as the description of session parameters (through the use of SDP) and the invocation of features and services. This means that the exchange of SIP signalling between an IP-PBX and the Proximus network shall serve one of the above purposes. In other words, SIP signalling shall be meaningful and the sending of useless SIP messages shall to be avoided !

Example of useless SIP signalling: A re-INVITE in an established session without any SDP included is often useless because typically re-INVITE is used to re-negotiate/change the session parameters. So, if there is no need to re-negotiate/change the session parameters no re-INVITE should be transmitted.

### 6.2. Registration/Authentication

IP-P(A)BX devices connected to the Proximus IMS network by use of Business Trunking or International Business Trunking **shall not** use registration. The identification and authentication of the IP-P(A)BX is done by making use of the one-to-one mapping of fixed IP-addresses. This is done through configuration in the Session Border Controller (SBC) of the Proximus IMS network.

Internal in the corporate network SIP end devices e.g. SIP phones, may register with the IP-P(A)BX itself. These registrations with the IP P(A)BX shall remain invisible to the Proximus IMS network.

Each IP-P(A)BX will be assigned one PBX public user identity (pbxPUID), aka Pilot ID aka PBXName. This is a non-dialable public identity.

An IP-P(A)BX has usually allocated to it one or more contiguous DN ranges.

Authentication procedures will not be requested for INVITE (session authentication).

### 6.3. Basic call

In Corporate VoIP several call types are possible:

- Private on-net calls are completely treated by the IP-P(A)BX and hence out of scope of this document.

- Public on-net calls are treated by the Proximus IMS network. This is a call between an IP-P(A)BX extension and an IMS user not part of the same IP-P(A)BX.
- Public off-net calls are treated by the Proximus IMS network. This is a call between an IP-P(A)BX extension and a non-IMS user (e.g. PSTN/ISDN, Mobile,...).

Basic call set-up and tear down complies with normal SIP behaviour as described in the relevant Standards and RFCs [4][9][10][11][13].

Basic call set-up in the SIP Business Trunking case happens without authentication. Since fixed IP addresses are used the Proximus IMS network identifies and authorizes the IP-P(A)BX based on the IP address.

Basic call set-up may use the mechanism for reliable transport of 1XX responses according to IETF RFC 3262 “Reliability of Provisional Responses in the Session Initiation Protocol (SIP)” [10], but it is not mandatory. The procedures for announcing the capability and use are described in [10].

Basic call set-up may use the mechanism for session timer according to IETF RFC 4028 “Session Timers in the Session Initiation Protocol (SIP)” [24], but it is not mandatory. The procedures for announcing the capability and use are described in [24].

The IP-P(A)BX shall use a Request-URI to originate a call as specified in PXM IMS Corporate VOIP - UNI specification - Address templates [40].

The IP-P(A)BX shall use the following own identity to originate a call:

- *From* header:
  - sip:Norm-SubA-dn@EnterpriseDomain;user=phone

The *From* header **shall** always contain a valid PUID, even when the CLIR service is to be invoked.

- *Contact* header
  - sip:Norm-SubA-dn@IP-addr-PBXName, or
  - sip:PBXName@IP-addr-PBXName

The IP address of the IP-P(A)BX in the *Contact* header **MUST** be the same as used in the configuration of the Proximus SBC because the Proximus IMS network uses this for admission control.

Remark: According to SIP Connect 1.1 [6], it is recommended that the IP-P(A)BX includes a *P-Asserted-ID* header in the *INVITE* request. The Proximus IMS network does not expect to receive this header and if received this header will be overwritten by the Proximus IMS network with the correct *P-Asserted-ID*.

Note: as “Norm-SubA-dn” any DN within the range of the IP-P(A)BX can be used.

### 6.3.1. Public on-net originating call set-up

Figure 1 shows the expected message flow for a public on-net originating call set-up.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

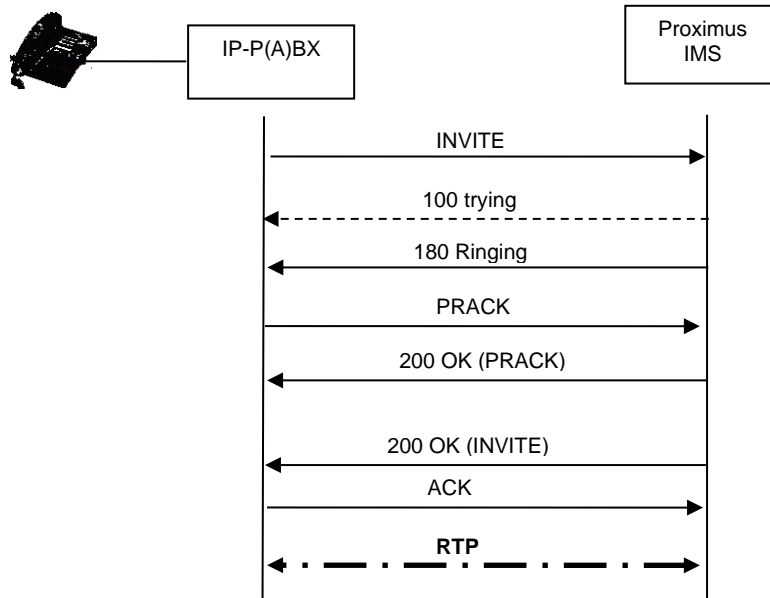


Figure 1: Basic call/session set-up (public on-net)

To set-up a public on-net call the IP-P(A)BX sends an *INVITE* message to the Proximus IMS network with the following headers:

- *Request-URI* as specified in PXM IMS Corporate VOIP - UNI specification - Address templates [40]
- *To* = <sip:Dialled-SubB-dn@ims.belgacom.be;user=phone>
- *From* = DisplayName <sip:Norm-SubA-dn@EnterpriseDomain;user=phone>
- *Via* = IP-addr-PBXName
- *Contact* = DisplayName <sip:Norm-SubA-dn@IP-addr-PBXName>

Note: The *DisplayName* in the *From* and *Contact* header is optional

#### Live example

```

Request-Line: INVITE sip:+3225016491@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-145103-5804
From: "Dory" <sip:+3225016490@dory.gent.be;user=phone>;tag=145103-86
To: <sip:+3225016491@ims.belgacom.be;user=phone>
Call-ID: 145103-6671
CSeq: 101 INVITE
User-Agent: Dory
Contact: "Dory" <sip:+3225016490@10.127.32.100:5060>
Expires: 15
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Supported: 100rel
    
```

Max-Forwards: 70  
Content-Type: application/sdp  
Content-Length: 252  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10000 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmtp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmtp:101 0-15  
Media Attribute (a):ptime:20  
Media Attribute (a): sendrecv

The *INVITE* contains SDP information regarding the proposed call parameters (e.g. codec, IP address and port number on which the IP-P(A)BX user wants to receive RTP).

The Proximus IMS network first returns a *100 Trying* response and performs normal call routing to the destination. When the destination is reached a *180 Ringing* response is returned to the IP-P(A)BX.

Remark: Because the initial *INVITE* sent by the IP-P(A)BX announced support for *100rel*, the Proximus IMS network decides to apply this mechanism. The *180 Ringing* response contains SDP information regarding the accepted call parameters and the following headers:

- *Require* = 100rel
- *Rseq* = <Rseqvalue>

#### Live example

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-145103-5804  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=145103-86  
To: <sip:+3225016491@ims.belgacom.be;user=phone>  
Call-ID: 145103-6671  
CSeq: 101 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 180 Ringing

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-150903-5952  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=150903-8  
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SDrskpc99-2115094522-1535375334492  
Call-ID: 145103-6671  
CSeq: 101 INVITE  
Contact: <sip:81.247.28.233:5060;transport=udp>  
**RSeq: 1036004910**  
**Require: 100rel**  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY  
Content-Type: application/sdp  
Content-Length: 191

Server: Alcatel-Lucent-HPSS/3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 152826899 1 IN IP4 81.247.28.233  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.28.233  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 14198 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15

The IP-P(A)BX shall send a *PRACK* request to the Proximus IMS network in order to acknowledge the receipt of the *180* response.

The *PRACK* request contains the following headers:

- *RAck* = <Rseqvalue as received> <RAckvalue> INVITE

#### Live example

Request-Line: *PRACK* sip:81.247.28.233:5060;transport=udp SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-145104-3108  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=145103-86  
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SDusidc99-1615397450-1535374254837  
Call-ID: 145103-6671  
CSeq: 102 *PRACK*  
**RAck: 1036004910 101 INVITE**  
Max-Forwards: 70  
Content-Length: 0

The Proximus IMS network shall send a *200 OK* response to the IP-P(A)BX in order to properly close the *PRACK* transaction.

#### Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-145104-3108  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=145103-86  
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SDusidc99-1615397450-1535374254837  
Call-ID: 145103-6671  
CSeq: 102 *PRACK*  
Server: Alcatel-Lucent-HPSS/3.0.3  
Content-Length: 0

When the call is answered the Proximus IMS network returns a *200 OK* response containing SDP information regarding the accepted call parameters (e.g. codec, IP address and port number on which the destination wants to receive RTP).

#### Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-145103-5804

From: <sip:+3225016490@dory.gent.be;user=phone>;tag=145103-86  
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SDusidc99-1615397450-1535374254837  
Call-ID: 145103-6671  
CSeq: 101 INVITE  
Contact: <sip:81.247.28.233:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml  
Content-Type: application/sdp  
Content-Length: 191  
Server: Alcatel-Lucent-HPSS/3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 152779878 1 IN IP4 81.247.28.233  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.28.233  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 14196 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15

The IP-P(A)BX sends an *ACK* message to the Proximus IMS network in order to properly close the *INVITE* transaction.

#### Live example

Request-Line: ACK sip:81.247.28.233:5060;transport=udp SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-150905-6332  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=150903-8  
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SDrskpc99-2115094522-1535375334492  
Call-ID: 145103-6671  
CSeq: 101 ACK  
Content-Length: 0

### 6.3.2. Public off-net originating call set-up

Figure 2 shows the expected message flow for a public off-net originating call set-up.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

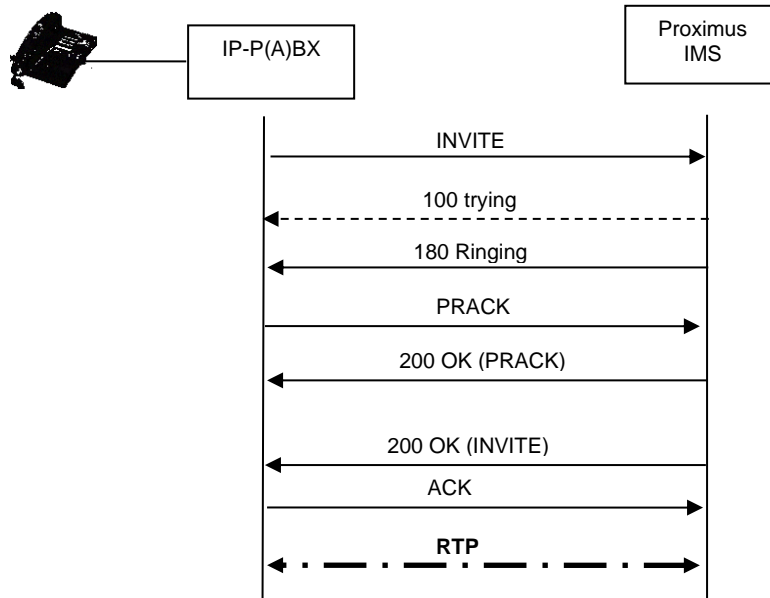


Figure 2: Basic call/session set-up (public off-net)

To set-up a public off-net call the IP-P(A)BX sends an *INVITE* message to the Proximus IMS network with the following headers:

- *Request-URI* as specified in PXM IMS Corporate VOIP - UNI specification - Address templates [40]
- *To* = <sip:Dialled-SubB-dn@ims.belgacom.be;user=phone>
- *From* = DisplayName <sip:Norm-SubA-dn@EnterpriseDomain;user=phone>
- *Via* = IP-addr-PBXName
- *Contact* = DisplayName <sip:Norm-SubA-dn@IP-addr-PBXName>

Note: The *DisplayName* in the *From*, *Contact* and *P-Asserted-ID* header is optional

[Live example](#)

```

Request-Line: INVITE sip:+32475339778@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-102608-133
From: "Dory" <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78
To: <sip:+32475339778@ims.belgacom.be;user=phone>
Call-ID: 102608-8109
CSeq: 102 INVITE
User-Agent: Dory
Contact: "Dory" <sip:+3225016490@10.127.32.100:5060>
Expires: 15
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Supported: 100rel
Max-Forwards: 70
    
```



Content-Type: application/sdp  
Content-Length: 252  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10000 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a):ptime:20  
Media Attribute (a): sendrecv

The *INVITE* contains SDP information regarding the proposed call parameters (e.g. codec, IP address and port number on which the IP-P(A)BX user wants to receive RTP).

The Proximus IMS network first returns a *100 Trying* response and performs normal call routing to the destination. When the destination is reached a *180 Ringing* response is returned to the IP-P(A)BX.

Remark: Because the initial *INVITE* sent by the IP-P(A)BX announced support for *100rel*, the Proximus IMS network decides to apply this mechanism. The *180 Ringing* response contains SDP information regarding the accepted call parameters and the following headers:

- *Require* = 100rel
- *Rseq* = <Rseqvalue>

#### Live example

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-102608-133  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78  
To: <sip:+32475339778@ims.belgacom.be;user=phone>  
Call-ID: 102608-8109  
CSeq: 102 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 183 Session Progress

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-102608-133  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78  
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDrck9c99-1335959284-1535444771530  
Call-ID: 102608-8109  
CSeq: 102 INVITE  
Contact: <sip:81.247.28.233:5060;transport=udp>  
RSeq: 1070723429  
Require: 100rel  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Content-Type: application/sdp  
Content-Length: 245  
Server: Alcatel-Lucent-HPSS/3.0.3

Session: Media  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 153663574 1 IN IP4 81.247.28.233  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.28.233  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 14224 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

The IP-P(A)BX shall send a *PRACK* request to the Proximus IMS network in order to acknowledge the receipt of the *180* response.

The *PRACK* request contains the following headers:

- *RAck* = <Rseqvalue as received> <RAckvalue> INVITE

Live example

Request-Line: *PRACK* sip:81.247.28.233:5060;transport=udp SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-145104-3108  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78  
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDrck9c99-1335959284-1535444771530  
Call-ID: 102608-8109  
CSeq: 103 *PRACK*  
**RAck: 1070723429 102 INVITE**  
Max-Forwards: 70  
Content-Length: 0

The Proximus IMS network shall send a *200 OK* response to the IP-P(A)BX in order to properly close the *PRACK* transaction.

Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-102611-2567  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78  
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDrck9c99-1335959284-1535444771530  
Call-ID: 102608-8109  
CSeq: 103 *PRACK*  
Server: Alcatel-Lucent-HPSS/3.0.3  
Content-Length: 0

When the call is answered the Proximus IMS network returns a *200 OK* response containing SDP information regarding the accepted call parameters (e.g. codec, IP address and port number on which the destination wants to receive RTP).

Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-102608-133  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78  
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDrck9c99-1335959284-1535444771530  
Call-ID: 102608-8109  
CSeq: 102 INVITE  
Contact: <sip:81.247.28.233:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml  
Content-Type: application/sdp  
Content-Length: 245  
Server: Alcatel-Lucent-HPSS/3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 153663574 1 IN IP4 81.247.28.233  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.28.233  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 14224 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

The IP-P(A)BX sends an ACK message to the Proximus IMS network in order to properly close the *INVITE* transaction.

[Live example](#)

Request-Line: ACK sip:81.247.28.233:5060;transport=udp SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-102617-9013  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78  
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDrck9c99-1335959284-1535444771530  
Call-ID: 102608-8109  
CSeq: 102 ACK  
Content-Length: 0

### 6.3.3. Public on-net terminating call set-up

Figure 3 shows the expected message flow for a public on-net terminating call set-up.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

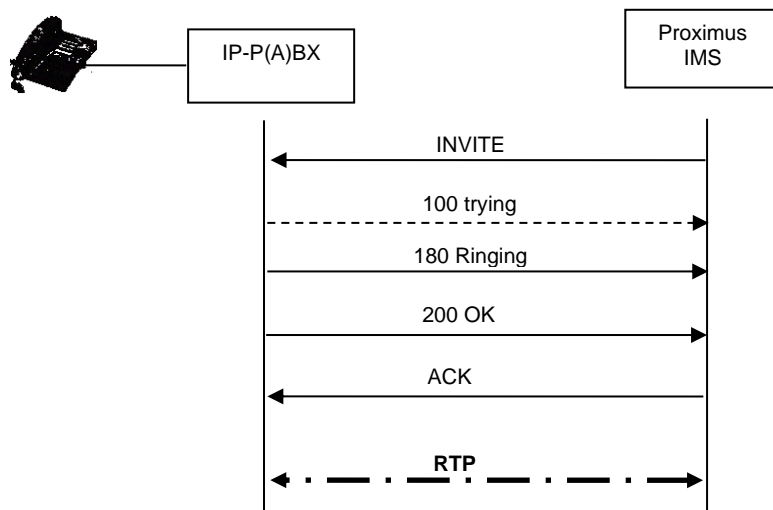


Figure 3: Basic call/session set-up (public on-net)

To set-up a call to the IP-P(A)BX, the Proximus IMS network sends an *INVITE* message to the IP-P(A)BX with the following headers:

- *Request-URI* = sip:Norm-SubB-dn@IP-addr-PBXName;user=phone
- *To* = <sip:Norm-SubB-dn@EnterpriseDomain;user=phone>
- *From* = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone>
- *Via* = IP-addr-IMS
- *Contact* = <sip:IP-addr-IMS>
- *P-Asserted-ID* = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone>
- *P-called-party-id* = <tel:Norm-SubB-dn>

Note 1: The *DisplayName* in the *From* header is optional

Note 2: The *P-asserted-id* header and *P-called-party-id* header are optional

Remark: Any incoming basic call to an IP-P(A)BX may have undergone diversion before it reaches the IP-P(A)BX. Therefore a diverting number can be present in the *INVITE* message in the *Diversion* header (the *Diversion Top* header contains the Redirecting Number, while the *Diversion Bottom* header contains the Original Called Number). The *Diversion* header has been documented in IETF RFC 5806 Diversion indication in SIP [31].

The *INVITE* contains SDP information regarding the proposed call parameters (e.g. codec, IP address and port number on which the Proximus IMS network wants to receive RTP).

Live example

```

Request-Line: INVITE sip:+3225016490@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKor7r0krkieinlo2ovc7j3lpa71
From: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>;
tag=SD8dg1a01-226252709-1535451702401-
To: "3225016490 ." <sip:+3225016490@ims.belgacom.be;user=phone>
    
```

Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043  
CSeq: 447041 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Privacy: none  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 193  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 153974618 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10250 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmtp:101 0-15  
Media Attribute (a): sendrecv

The Proximus IMS network expects a *100 Trying* response, followed by a *180 Ringing*, when the destination user is reached.

#### Live example

Status-Line: SIP/2.0 100 Trying  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKor7rokrkieinlo2ovc7j3lpa71  
From: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SD8dg1a01-226252709-1535451702401-  
To: "3225016490 ." <sip:+3225016490@ims.belgacom.be;user=phone>  
Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043  
CSeq: 447041 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 180 Ringing  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKor7rokrkieinlo2ovc7j3lpa71  
From: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SD8dg1a01-226252709-1535451702401-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=122143-97  
Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043  
CSeq: 447041 INVITE  
Contact: <sip:+3225016491@10.127.32.100:5061>  
Content-Length: 0

When the call is answered the Proximus IMS network expects a *200 OK* response containing SDP information regarding the accepted call parameters (e.g. codec, IP address and port number on which the destination wants to receive RTP).

#### Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKor7rokrkieinlo2ovc7j3lpa71  
From: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SD8dg1a01-226252709-1535451702401-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=122143-97  
Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043  
CSeq: 447041 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016491@10.127.32.100:5061>  
Content-Type: application/sdp  
Content-Length: 193  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10102 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): sendrecv

The Proximus IMS network sends an ACK message to the IP-P(A)BX in order to properly close the *INVITE* transaction.

#### Live example

Request-Line: ACK sip:+3225016491@10.127.32.100:5061;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKc38t852eoltmkaql9v4b91vui7-g14t0h5  
From: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SD8dg1a01-226252709-1535451702401-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=122143-97  
Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043  
CSeq: 447041 ACK  
Contact: <sip:81.247.255.118:5060>  
Max-Forwards: 68  
Content-Length: 0  
User-Agent: Alcatel-Lucent-HPSS v3.0.3

### 6.3.4. Public off-net terminating call set-up

Figure 4 shows the expected message flow for a public off-net terminating call set-up.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

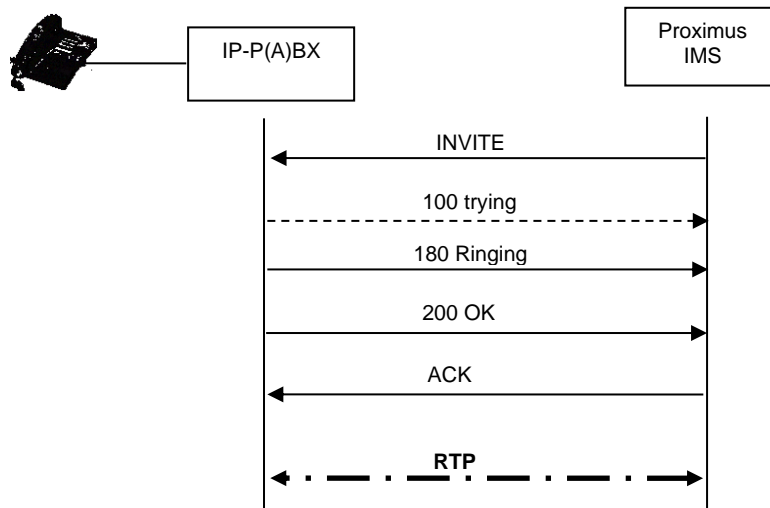


Figure 4: Basic call/session set-up (public off-net)

To set-up a call to the IP-P(A)BX, the Proximus IMS network sends an *INVITE* message to the IP-P(A)BX with the following headers:

- *Request-URI* = sip:Norm-SubB-dn@IP-addr-PBXName;user=phone
- *To* = <sip:Norm-SubB-dn@EnterpriseDomain;user=phone>
- *From* = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone>
- *Via* = IP-addr-IMS
- *Contact* = <sip:IP-addr-IMS>
- *P-asserted-id* = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone> or <tel:Norm-SubA-dn>
- *P-called-party-id* = <tel:Norm-SubB-dn>

Note 1: The *DisplayName* in the *From* header is optional

Note 2: The *P-asserted-id* header and *P-called-party-id* header are optional

Remark: Any incoming basic call to an IP-P(A)BX may have undergone diversion before it reaches the IP-P(A)BX. Therefore, a diverting number can be present in the *INVITE* message in the *Diversion* header (the *Diversion Top* header contains the Redirecting Number, while the *Diversion Bottom* header contains the Original Called Number). The *Diversion* header has been documented in IETF RFC 5806 Diversion indication in SIP [31].

The *INVITE* contains SDP information regarding the proposed call parameters (e.g. codec, IP address and port number on which the Proximus IMS network wants to receive RTP).

#### Live example

**Request-Line: INVITE sip:+3225016490@ims.belgacom.be;user=phone SIP/2.0**  
**Message Header**  
**Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKmogsekqqlch1bqc7mpf6vsa20**  
**From: <sip:+32475339778@fed.ipcx.be:51007;user=phone>;tag=SDg5hja01-1550815237-1535454301808-**

**To: "3225016490 ."<sip:+3225016490@ims.belgacom.be;user=phone>**  
Call-ID: SDg5hja01-fe05be9c4e355294e5f427599fd6d85d-aobq043  
CSeq: 1746745 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Privacy: none  
Supported: 100rel  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept:  
application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,multipart/mixed

Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 294  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info

Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 154030154 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10252 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

The Proximus IMS network expects a *100 Trying* response, followed by a *180 Ringing*, when the destination user is reached.

#### Live example

Status-Line: SIP/2.0 100 Trying  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKmogsekqqlch1bqc7mpf6vsa20  
From: <sip:+32475339778@fed.ipcx.be:51007;user=phone>;tag=SDg5hja01-1550815237-1535454301808-  
To: "3225016490 ."<sip:+3225016490@ims.belgacom.be;user=phone>  
Call-ID: SDg5hja01-fe05be9c4e355294e5f427599fd6d85d-aobq043  
CSeq: 1746745 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 180 Ringing  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKmogsekqqlch1bqc7mpf6vsa20  
From: <sip:+32475339778@fed.ipcx.be:51007;user=phone>;tag=SDg5hja01-1550815237-1535454301808-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=130502-12  
Call-ID: SDg5hja01-fe05be9c4e355294e5f427599fd6d85d-aobq043  
CSeq: 1746745 INVITE  
Contact: <sip:+3225016491@10.127.32.100:5061>  
Content-Length: 0



When the call is answered the Proximus IMS network expects a 200 OK response containing SDP information regarding the accepted call parameters (e.g. codec, IP address and port number on which the destination wants to receive RTP).

Live example

```
Status-Line: SIP/2.0 200 OK
Message Header
  Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKmogsekqqlch1bqc7mpf6vsa20
  From: <sip:+32475339778@fed.ipcx.be:51007;user=phone>;tag=SDg5hja01-1550815237-1535454301808-
  To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=130502-12
  Call-ID: SDg5hja01-fe05be9c4e355294e5f427599fd6d85d-aobq043
  CSeq: 1746745 INVITE
  User-Agent: Dory
  Contact: <sip:+3225016491@10.127.32.100:5061>
  Content-Type: application/sdp
  Content-Length: 193
  Accept: application/sdp
Message Body
Session Description Protocol
  Session Description Protocol Version (v): 0
  Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100
  Session Name (s): SIP Call
  Connection Information (c): IN IP4 10.127.32.100
  Time Description, active time (t): 0 0
  Media Description, name and address (m): audio 10104 RTP/AVP 8 101
  Media Attribute (a): rtpmap:8 PCMA/8000
  Media Attribute (a): rtpmap:101 telephone-event/8000
  Media Attribute (a): fmtp:101 0-15
  Media Attribute (a): sendrecv
```

The Proximus IMS network sends an ACK message to the IP-P(A)BX in order to properly close the *INVITE* transaction.

Live example

```
Request-Line: ACK sip:+3225016491@10.127.32.100:5061;user=phone SIP/2.0
Message Header
  Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK8omdsrvdauon87lioub8667ru6-g6eqsg5
  From: <sip:+32475339778@fed.ipcx.be:51007;user=phone>;tag=SDg5hja01-1550815237-1535454301808-
  To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=130502-12
  Call-ID: SDg5hja01-fe05be9c4e355294e5f427599fd6d85d-aobq043
  CSeq: 1746745 ACK
  Contact: <sip:81.247.255.118:5060>
  Max-Forwards: 68
  Content-Length: 0
  User-Agent: Alcatel-Lucent-HPSS v3.0.3
```

### 6.3.5. Forward call tear down

Figure 5 shows the expected message flow for a call tear down initiated by the IP-P(A)BX user.

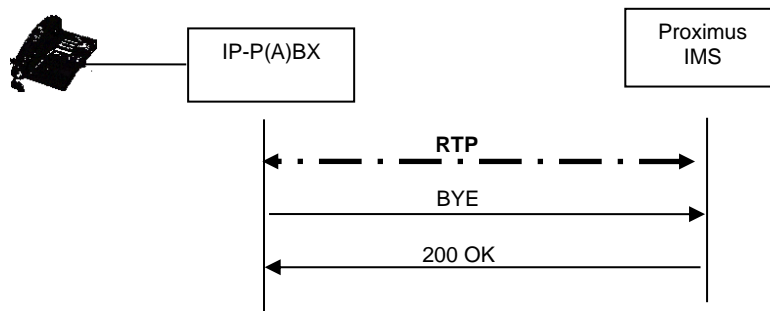


Figure 5: Basic call/session forward teardown

To end a public on-net or a public off-net call the IP-P(A)BX sends a *BYE* message to the Proximus IMS network.

- *Call-ID* = matches the *Call-ID* of the concerned call
- *CSeq* = is incremented with regard to the previous transaction related to this call

#### Live example

```

Request-Line: BYE sip:81.247.255.118:5060 SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-122202-4197
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=122143-97
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SD8dg1a01-226252709-1535451702401-
Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043
CSeq: 447042 BYE
Content-Length: 0
    
```

The Proximus IMS network returns a *200 OK* response.

#### Live example

```

Status-Line: SIP/2.0 200 OK
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-122202-4197
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=122143-97
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=SD8dg1a01-226252709-1535451702401-
Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043
CSeq: 447042 BYE
Server: Alcatel-Lucent-HPSS/3.0.3
Content-Length: 0
    
```

### 6.3.6. Backward call tear down

Figure 6 shows the expected message flow for a call tear down initiated by the Proximus IMS network.

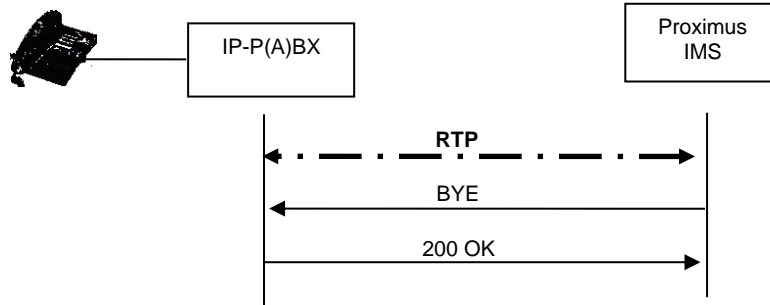


Figure 6: Basic call/session backward teardown

To end a public on-net or a public off-net call the Proximus IMS network sends a *BYE* message to the IP-P(A)BX.

- *Call-ID* = matches the *Call-ID* of the concerned call
- *CSeq* = is incremented with regard to the previous transaction related to this call

#### Live example

Request-Line: *BYE sip:+3225016490@10.127.32.100:5060;user=phone SIP/2.0*

Message Header

Via: *SIP/2.0/UDP 81.247.28.233:5060;branch=z9hG4bKl7sff683kht6pe4eof91t9er05-v9kneksa*

From: *<sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDrck9c99-1335959284-1535444771530*

To: *<sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78*

Call-ID: *102608-8109*

CSeq: *1070721832 BYE*

Max-Forwards: *68*

Content-Length: *0*

User-Agent: *Alcatel-Lucent-HPSS v3.0.3*

The IP-P(A)BX returns a *200 OK* response.

#### Live example

Status-Line: *SIP/2.0 200 OK*

Message Header

Via: *SIP/2.0/UDP 81.247.28.233:5060;branch=z9hG4bKl7sff683kht6pe4eof91t9er05-v9kneksa*

From: *<sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDrck9c99-1335959284-1535444771530*

To: *<sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78*

Call-ID: *102608-8109*

CSeq: *1070721832 BYE*

Content-Length: *0*

## 6.4. Supplementary services

All supplementary services shall be executed at -and managed by- the IP-P(A)BX. There is no individual user profile in the Proximus IMS network for the extensions behind the IP-P(A)BX.

Services executed by the IP-P(A)BX and involving internal users only shall not be visible for the Proximus IMS network.

Services executed by the IP-P(A)BX involving external users need particular interaction with the Proximus IMS network. The expected behaviour is discussed in the following clauses.

### 6.4.1. Calling Line Identity Presentation (CLIP)

The Calling Line Identification Presentation service is enabled by default and applies to all PBX iDN and to the pbxPUIID.

Note: the pbxPUIID is in fact a non-dialable public identity and hence in practice it will not be used as CLI

#### 6.4.1.1. Incoming call to the IP-P(A)BX

The CLIP service sends the identity of the calling line to the IP-P(A)BX. The calling line identity can be provided by the network or it can include identity information supplied by the caller and validated by the network.

The calling line identity can be contained in the *From* and *P-Asserted-Id* headers in the *INVITE* message sent to the IP-P(A)BX. The presentation information of the calling line identity, which establishes the identity type (Presentation Allowed or Presentation Restricted) is contained in the *Privacy* header in the *INVITE* message.

The format of the calling line identity information in the *From* and *P-Asserted-Id* headers in the *INVITE* message is:

- *From* = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone>
- *P-Asserted-ID* = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone> or <tel:Norm-SubA-dn>

Note 1: The *DisplayName* in the *From* header and *P-Asserted-ID* header is optional

Note 2: The Norm-SubA-dn in the *From* header and *P-Asserted-ID* header is not necessarily the same. For example in case of interworking with legacy ISDN two calling line identities may be delivered.

The *Privacy* header in the *INVITE* message has the value "none". The absence of the *Privacy* header equally means "no privacy".

#### Live example

```
Request-Line: INVITE sip:+3225016490@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKor7rokrkieinlo2ovc7j3lpa71
From: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>;
tag=SD8dg1a01-226252709-1535451702401-
To: "3225016490 ." <sip:+3225016490@ims.belgacom.be;user=phone>
Call-ID: SD8dg1a01-a669035ffc02cfb7867ff0079247dab8-aobq043
CSeq: 447041 INVITE
Contact: <sip:81.247.255.118:5060;transport=udp>
Privacy: none
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
```

Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 193  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 153974618 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10250 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmtp:101 0-15  
Media Attribute (a): sendrecv

#### 6.4.1.2. Outgoing call from the IP-P(A)BX

The calling line identity shall be contained in the *From* header in the *INVITE* message sent by the IP-P(A)BX.

- *From* = DisplayName <sip:Norm-SubA-dn@EnterpriseDomain;user=phone>

Note: The *DisplayName* in the *From* header is optional

[Live example](#)

##### Session Initiation Protocol (INVITE)

Request-Line: INVITE sip:+32475339778@ims.belgacom.be;user=phone SIP/2.0

##### Message Header

Via: SIP/2.0/UDP 10.127.32.100:5060;branch=z9hG4bK-102608-133

**From:** "Dory" <sip:+3225016490@dory.gent.be;user=phone>;tag=102608-78

To: <sip:+32475339778@ims.belgacom.be;user=phone>

Call-ID: 102608-8109

CSeq: 102 INVITE

User-Agent: Dory

Contact: "Dory" <sip:+3225016490@10.127.32.100:5060>

Expires: 15

Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE

Max-Forwards: 70

Content-Type: application/sdp

Content-Length: 252

Accept: application/sdp

##### Message Body

##### Session Description Protocol

Session Description Protocol Version (v): 0

Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100

Session Name (s): SIP Call

Connection Information (c): IN IP4 10.127.32.100

Time Description, active time (t): 0 0

Media Description, name and address (m): audio 10000 RTP/AVP 8 18 101

Media Attribute (a): rtpmap:8 PCMA/8000

Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): ptime:20  
Media Attribute (a): sendrecv

Note: According to SIP Connect 1.1 [6], it is recommended that the IP-P(A)BX includes a *P-Asserted-ID* header in the *INVITE* request. The Proximus IMS network does not expect to receive this header and if received this header will be overwritten by the Proximus IMS network with the correct *P-Asserted-ID*.

## 6.4.2. Calling Line Identity presentation Restriction (CLIR)

The Calling Line Identity presentation Restriction (CLIR) service can be invoked on a call by call basis per individual number.

### 6.4.2.1. Incoming call to the IP-P(A)BX

In case the CLIR service has been invoked by the calling user of an incoming call to the IP-P(A)BX, the *From* header in the *INVITE* will not contain the identity information of the calling user and the *P-Asserted-Id* header will not be present in the *INVITE*. The presentation information of the calling line identity, which establishes the identity type (Presentation Allowed or Presentation Restricted) is contained in the *Privacy* header in the *INVITE* message.

- *From* = DisplayName <sip:Anonymous@anonymous.invalid>
- *Privacy* = id and/or header and/or User

Note: if present *DisplayName* will have the value “Anonymous”.

### 6.4.2.2. Outgoing call from the IP-P(A)BX

In order to prevent the presentation of the calling user’s identity (invoke CLIR) the IP-P(A)BX shall include a *Privacy* header in the *INVITE*. The value of the *Privacy* header in the *INVITE* shall be *id* and/or *header* and/or *User*

In any case the *From* header **shall** always contain a PUID, even when the CLIR service is to be invoked.

- *From* = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone>
- *Privacy* = id and/or header and/or User

Note 1: the values) of the *Privacy* header shall be chosen in accordance with RFC 3323 [16] and RFC 3325 [17].

Note 2: for the invocation of the CLIR service it is recommended to use at least the *Privacy* value “user”.

## 6.4.3. Call Forwarding Services

The call forwarding feature(s) shall be managed and handled by the IP-P(A)BX.

The following behaviour is valid for all flavours (e.g. unconditional (CFU), on busy (CFB), on no reply (CFNR), ...) of call forwarding executed by the IP-P(A)BX.

In case an incoming call to the IP-P(A)BX is forwarded to a destination **external** to the IP-P(A)BX, the IP-P(A)BX shall:

- either create the forwarding by setting-up a new (forwarded) call. The incoming call and the outgoing (forwarded) call are considered being 2 separate calls. A *Diversion* header may be included in the outgoing (forwarded) call in which case the calling line identity possibly shown to the forwarded-to destination will be the original calling line identity, else the calling line identity possibly shown to the forwarded-to destination will be the identity of the forwarding IP-P(A)BX user (or the IP-P(A)BX general number). The *History-info* header **MUST NOT** be included in the outgoing (forwarded) call.
- or return a 302 Moved Temporarily response containing the forwarded-to-number in the *Contact* header. The Proximus IMS network will then redirect the call to the forwarded-to-number. The calling line identity possibly shown to the forwarded-to destination will be the identity of the original calling user.

### 6.4.3.1. Call forward via new (outgoing) call

Figure 7 shows the expected message flow for a call forward via a new call.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

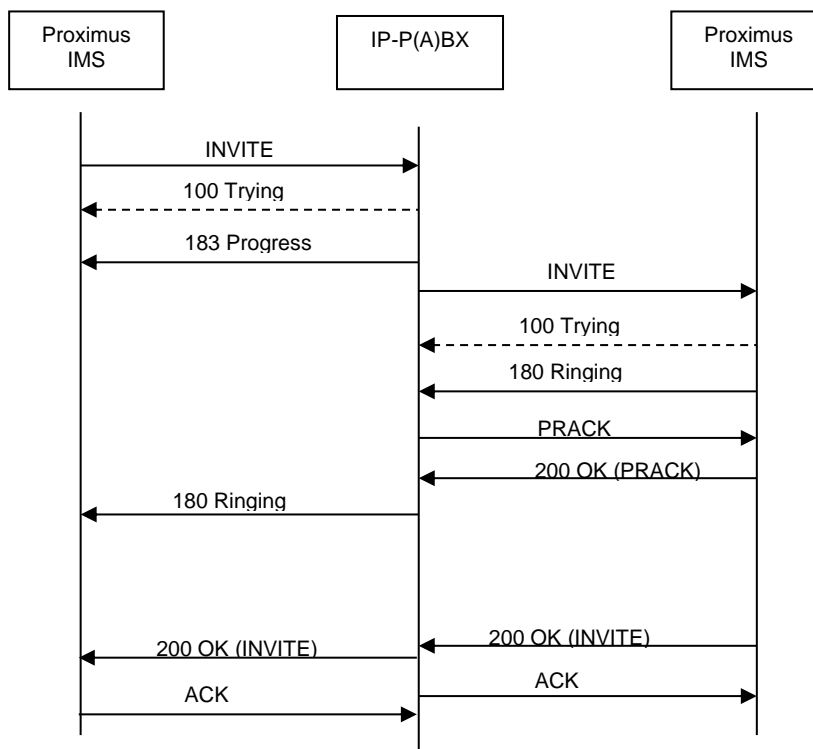


Figure 7: Call forward via new outgoing call

In order to set-up a call to an IP-P(A)BX the Proximus IMS network sends an *INVITE* message to the IP-P(A)BX with:

- *From* = DisplayName <sjp:Norm-SubA-dn@ims.belgacom.be;user=phone>
- *To* = DisplayName <sjp:Norm-SubB-dn@EnterpriseDomain;user=phone>

The IP-P(A)BX returns a *100 Trying* response.

Live example

```
Request-Line: INVITE sip:+3225016491@ims.belgacom.be;user=phone SIP/2.0
Message Header
  Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK6u6tulpk0300hthg64kho693p2
  From: "+3225016490"<sip:+3225016490@ims.belgacom.be>;tag=SDibi7b01-153298320-1536308540522-
  To: "3225016491."<sip:+3225016491@ims.belgacom.be;user=phone>
  Call-ID: SDibi7b01-d310216f3efce39666d4de328ed86b11-aobq043
  CSeq: 428866102 INVITE
  Contact: <sip:81.247.255.118:5060;transport=udp>
  Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
  Accept: application/media_control+xml,application/sdp,application/vnd.etsi.pstn+xml,
multipart/mixed
  Max-Forwards: 68
  Content-Type: application/sdp
  Content-Length: 267
  User-Agent: Alcatel-Lucent-HPSS v3.0.3
  Recv-Info: x-broadworks-client-session-info
Message Body
  Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): BroadWorks 166747656 1 IN IP4 81.247.255.118
    Session Name (s): -
    Connection Information (c): IN IP4 81.247.255.118
    Time Description, active time (t): 0 0
    Media Description, name and address (m): audio 13454 RTP/AVP 8 18 101
    Media Attribute (a): rtpmap:8 PCMA/8000
    Media Attribute (a): rtpmap:18 G729/8000
    Media Attribute (a): fmp:18 annexb=no
    Media Attribute (a): rtpmap:101 telephone-event/8000
    Media Attribute (a): fmp:101 0-16
    Media Attribute (a): silenceSupp:off - - - -
```

```
Status-Line: SIP/2.0 100 Trying
Message Header
  Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK6u6tulpk0300hthg64kho693p2
  From: "+3225016490"<sip: +3225016490@ims.belgacom.be>;tag=SDibi7b01-153298320-1536308540522-
  To: "3225016491."<sip:+3225016491@ims.belgacom.be;user=phone>
  Call-ID: SDibi7b01-d310216f3efce39666d4de328ed86b11-aobq043
  CSeq: 428866102 INVITE
  Content-Length: 0
```

When the IP-P(A)BX invokes the call forward feature, a *183 Progress* response is sent to the Proximus IMS network.

Live example

```
Status-Line: SIP/2.0 183 Session Progress
Message Header
  Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK6u6tulpk0300hthg64kho693p2
  From: "+3225016490"<sip: +3225016490@ims.belgacom.be>;tag=SDibi7b01-153298320-1536308540522-
  To: "3225016491."<sip:+3225016491@ims.belgacom.be;user=phone>
```



Call-ID: SDibi7bo1-d310216f3efce39666d4de328ed86b11-aobq043  
CSeq: 428866102 INVITE  
Content-Length: 0

The IP-P(A)BX starts a new call to the “forwarded-to-destination” by sending an *INVITE* message to the Proximus IMS network with:

- From = DisplayName <sip:Norm-SubB-dn@EnterpriseDomain;user=phone>
- To = <sip:Dialled-SubC-dn@EnterpriseDomain;user=phone>

#### Live example

Request-Line: INVITE sip:+32475339778@ims.belgacom.be;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-102224-5898  
**From: "+3225016490" <sip:+3225016490@ims.belgacom.be>;tag=102224-2**  
**To: <sip:+32475339778@ims.belgacom.be;user=phone>**  
Call-ID: 102224-2159  
CSeq: 102 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016491@10.127.32.100:5061>  
Expires: 15  
Allow: CANCEL,BYE,INVITE, ACK  
supported: 100rel  
Diversion: "Dory" <sip:+3225016491@dory.gent.be;user=phone>  
Content-Type: application/sdp  
Content-Length: 193  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10000 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): sendrecv

The Proximus IMS network returns a *100 Trying* response to the IP-P(A)BX.

#### Live example

Status-Line: SIP/2.0 100 Trying  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-102224-5898  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=102224-2  
To: <sip:+32475339778@ims.belgacom.be;user=phone>  
Call-ID: 102224-2159  
CSeq: 102 INVITE  
Content-Length: 0

When the “forwarded-to-destination” is reached the Proximus IMS network returns a *180 Ringing* response to the IP-P(A)BX.

In this case the *180 Ringing* response contains a *Require* header with value *100rel*, which means that reliable transport of *1XX* responses is required. Consequently the IP-P(A)BX must send a *PRACK* message to the Proximus IMS network in order to confirm the receipt of the *180 Ringing* response..

[Live example](#)

```
Status-Line: SIP/2.0 183 Session Progress
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-102224-5898
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=102224-2
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDncjbc99-1078179111-1536308547508
Call-ID: 102224-2159
CSeq: 102 INVITE
Contact: <sip:81.247.255.118:5060;transport=udp>
RSeq: 428869594
P-Asserted-Identity: <sip:+32475339778@ims.belgacom.be;user=phone>
Privacy: none
Require: 100rel
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Content-Type: application/sdp
Content-Length: 247
Server: Alcatel-Lucent-HPSS/3.0.3
Session: Media
Message Body
Session Description Protocol
  Session Description Protocol Version (v): 0
  Owner/Creator, Session Id (o): BroadWorks 166747811 1 IN IP4 81.247.255.118
  Session Name (s): -
  Connection Information (c): IN IP4 81.247.255.118
  Time Description, active time (t): 0 0
  Media Description, name and address (m): audio 13456 RTP/AVP 8 101
  Media Attribute (a): rtpmap:8 PCMA/8000
  Media Attribute (a): rtpmap:101 telephone-event/8000
  Media Attribute (a): fmp:101 0-15
  Media Attribute (a): maxptime:30
  Media Attribute (a): silenceSupp:off - - - -
  Media Attribute (a): ptime:20
  Media Attribute (a): sendrecv
```

```
Request-Line: PRACK sip:81.247.255.118:5060;transport=udp SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-102227-7351
From: <sip:+3225016490@ims.belgacom.be>;tag=102224-2
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDncjbc99-1078179111-1536308547508
Call-ID: 102224-2159
CSeq: 103 PRACK
RAck: 428869594 102 INVITE
Max-Forwards: 70
Content-Length: 0
```

The Proximus IMS network returns a *200 OK* response to the IP-P()BX in order to properly close the *PRACK* transaction.

[Live example](#)

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-102227-7351  
From: <sip:+3225016490@ims.belgacom.be>;tag=102224-2  
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDncjbc99-1078179111-1536308547508  
Call-ID: 102224-2159  
CSeq: 103 PRACK  
Server: Alcatel-Lucent-HPSS/3.0.3  
Content-Length: 0

After the successful *PRACK* transaction the IP-P(A)BX returns a *180 Ringing* response to the Proximus IMS network for the original incoming *INVITE* message.

Live example

Status-Line: SIP/2.0 180 Ringing  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK6u6tulpk03oohtg64kho693p2  
From: <sip: +3225016490@ims.belgacom.be>;tag=SDibi7b01-153298320-1536308540522-  
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=102228-55  
Call-ID: SDibi7b01-d310216f3efce39666d4de328ed86b11-aobq043  
CSeq: 428866102 INVITE  
Contact: <sip:+3225016491@10.127.32.100:5061>  
Content-Length: 0

When the “forwarded-to-destination” answers the call, the Proximus IMS network sends a *200 OK* response to the IP-P(A)BX.

Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-102224-5898  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=102224-2  
To: <sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDncjbc99-1078179111-1536308547508  
Call-ID: 102224-2159  
CSeq: 102 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
P-Asserted-Identity: <sip:+32475339778@fed.ipcx.be;user=phone>  
Privacy: none  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml  
Content-Type: application/sdp  
Content-Length: 247  
Server: Alcatel-Lucent-HPSS/3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 166747811 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13456 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000

Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20  
Media Attribute (a): sendrecv

The IP-P(A)BX returns an *ACK* message in order to properly close the *INVITE* transaction of the call to the "forwarded-to-destination".

Live example

Request-Line: *ACK sip:81.247.255.118:5060;transport=udp SIP/2.0*  
Message Header  
Via: *SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-102224-5898*  
From: *<sip:+3225016490@ims.belgacom.be;user=phone>;tag=102224-2*  
To: *<sip:+32475339778@ims.belgacom.be;user=phone>;tag=SDncjbc99-1078179111-1536308547508*  
Call-ID: *102224-2159*  
CSeq: *102 ACK*  
Content-Length: *0*

The IP-P(A)BX sends a *200 OK* response to the Proximus IMS network for the original *INVITE* transaction.

Live example

Status-Line: *SIP/2.0 200 OK*  
Message Header  
Via: *SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK6u6tulpk0300hthg64kho693p2*  
From: *" +322501690" <sip: +3225016490@ims.belgacom.be>;tag=SDibi7b01-153298320-1536308540522-*  
To: *"3225016491." <sip:+3225016491@ims.belgacom.be;user=phone>*  
Call-ID: *SDibi7b01-d310216f3efce39666d4de328ed86b11-aobq043*  
CSeq: *428866102 INVITE*  
Contact: *<sip:+325016491@10.127.32.100:5060>*  
Content-Type: *application/sdp*  
Content-Length: *193*  
Accept: *application/sdp*  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): *0*  
Owner/Creator, Session Id (o): *Dory 27106 3566 IN IP4 10.127.32.100*  
Session Name (s): *SIP Call*  
Connection Information (c): *IN IP4 10.127.32.100*  
Time Description, active time (t): *0 0*  
Media Description, name and address (m): *audio 10000 RTP/AVP 8 101*  
Media Attribute (a): *rtpmap:8 PCMA/8000*  
Media Attribute (a): *rtpmap:101 telephone-event/8000*  
Media Attribute (a): *fmp:101 0-15*  
Media Attribute (a): *sendrecv*

The Proximus IMS network returns an *ACK* message in order to properly close the *INVITE* transaction of the original call.

Live example

Request-Line: *ACK sip:+3225016491@10.127.32.100:5061;user=phone SIP/2.0*  
Message Header  
Via: *SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK7khi8saocuovl66gujq8r66ca3-pg9p8nh5*

From: "+3225016490" <sip: +3225016490@ims.belgacom.be>; tag=SDibi7b01-153298320-1536308540522-  
 To: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>  
 Call-ID: SDibi7b01-d310216f3efce39666d4de328ed86b11-aobq043  
 CSeq: 428866102 ACK  
 Contact: <sip:81.247.255.118:5060>  
 Max-Forwards: 68  
 Content-Length: 0  
 User-Agent: Alcatel-Lucent-HPSS v3.0.3

### 6.4.3.2. Call forward via 302 Moved temporary response

Figure 8 shows the expected message flow for a call forward via a 302 Moved temporarily response.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

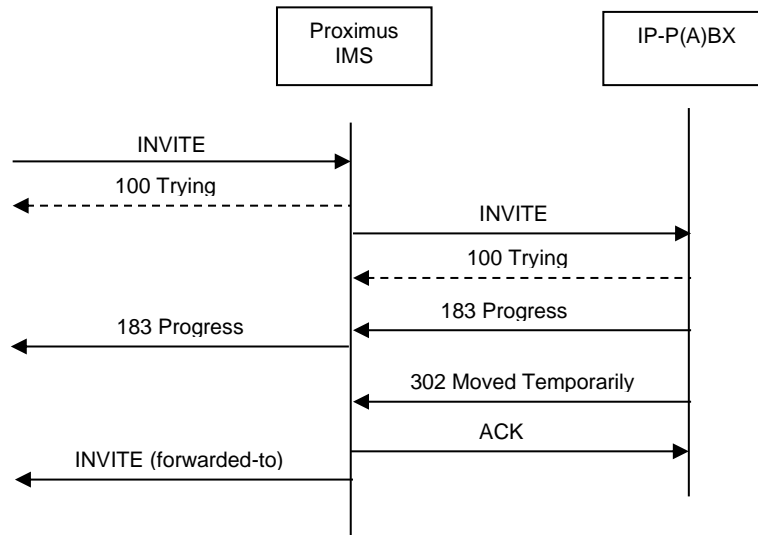


Figure 8: Call forward via 302 moved temporarily

In order to set-up a call to an IP-P(A)BX the Proximus IMS network sends an *INVITE* message to the IP-P(A)BX with:

- From = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone>
- To = DisplayName <sip:Norm-SubB-dn@ims.belgacom.be;user=phone>

The IP-P(A)BX returns a *100 Trying* response.

#### Live example

```

Request-Line: INVITE sip:+3225016491@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK24r6ft6jgdk5va4erqg5cm2c90
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=SDqkdve01-1644888295-1536313464920-
To: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>
Call-ID: SDqkdve01-9f943f9ac622ofe84cbf2f1e3dc4140c-aobq043
  
```

CSeq: 431328301 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 267  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 166994930 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13474 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-16  
Media Attribute (a): silenceSupp:off - - -

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK24r6ft6jgdk5va4erqg5cm2c90  
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=SDqkdve01-1644888295-  
1536313464920-  
To: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>  
Call-ID: SDqkdve01-9f943f9ac622ofe84cbf2f1e3dc4140c-aobq043  
CSeq: 431328301 INVITE  
Content-Length: 0

When the IP-P(A)BX invokes the call forward feature, a *183 Progress* response is sent to the Proximus IMS network, followed by a *302 Moved temporarily* response containing:

- Contact = <sip:Dialled-SubC-dn@ims.belgacom.be;user=phone>

#### Live example

Status-Line: SIP/2.0 183 Session Progress

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK24r6ft6jgdk5va4erqg5cm2c90  
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=SDqkdve01-1644888295-  
1536313464920-  
To: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>  
Call-ID: SDqkdve01-9f943f9ac622ofe84cbf2f1e3dc4140c-aobq043  
CSeq: 431328301 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 302 Moved Temporarily

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK24r6ft6jgdk5va4erqg5cm2c90  
From: <sip:anonymous@anonymous.invalid>;tag=SDqkdve01-1644888295-  
1536313464920-

To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=114425-73  
Call-ID: SDqkdve01-9f943f9ac6220fe84cbf2f1e3dc4140c-aobq043  
CSeq: 431328301 INVITE  
**Contact: <sip:+32475339778@ims.belgacom.be;user=phone>**  
Content-Length: 0

The Proximus IMS network returns an ACK message in order to properly close the INVITE transaction.

Live example

Request-Line: ACK sip:+3225016491@ims.belgacom.be;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK24r6ft6jgdk5va4erqg5cm2c90  
From: <sip:anonymous@anonymous.invalid>;tag=SDqkdve01-1644888295-1536313464920-  
To: <sip:+3225016491@ims.belgacom.be;user=phone>;tag=114425-73  
Call-ID: SDqkdve01-9f943f9ac6220fe84cbf2f1e3dc4140c-aobq043  
CSeq: 431328301 ACK  
Max-Forwards: 68  
Content-Length: 0

The Proximus IMS network starts a new call towards the forwarded-to-destination received in the Contact header of the 302 Moved temporarily response:

- Request-URI = <sip:Norm-SubC-dn@ims.belgacom.be;user=phone>
- From = DisplayName <sip:Norm-SubA-dn@ims.belgacom.be;user=phone>
- To = DisplayName <sip:Norm-SubC-dn@ims.belgacom.be;user=phone>
- Diversion = DisplayName <sip:Norm-SubB-dn@ims.belgacom.be;user=phone>

Live example

Request-Line: INVITE **sip:+32475339778@ims.belgacom.be;user=phone** SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.48.46.39;branch=z9hG4bKBroadWorks.-1pfarso-10.48.96.212V5060-0-431328590-2036844104-1536313465499-  
**From: <sip:anonymous@anonymous.invalid>;tag=2036844104-1536313465499-**  
**To: <sip:+32475339778@ims.belgacom.be;user=phone>**  
Call-ID: BW0944254990709182028000234@10.48.46.39  
CSeq: 431328590 INVITE  
Contact: <sip:+3225016490;... >  
P-Asserted-Identity: "... " <sip:... @ims.belgacom.be>, "... " <tel:+3225016328>  
Privacy: user;id;critical  
**Diversion: "3225016491 ." <sip:+3225016491@ims.belgacom.be;user=phone>;**  
privacy=off;reason=unknown;counter=1  
Route: ...  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY  
  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,application/x-broadworks-call-center+xml,multipart/mixed  
Supported:  
Max-Forwards: 70  
Content-Type: application/sdp  
Content-Length: 263  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 166994956 1 IN IP4 10.52.47.135  
Session Name (s): -

Connection Information (c): IN IP4 10.47.14.197  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 11060 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmtp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmtp:101 0-16  
Media Attribute (a): silenceSupp:off - - - -

The remainder of the message sequence is like any other normal basic call.

#### 6.4.4. Outgoing Call Barring

In case the IP-P(A)BX applies Outgoing Call Barring to one of his users, the call shall **not** be sent to the Proximus IMS network.

Additionally, a network Outgoing Call Barring can be programmed for the complete IP-P(A)BX and/or for each iDN. This OCB is offered on a permanent basis only and can not be activated or de-activated via a dialled command.

In case a IP-P(A)BX user tries to make a call to a barred destination, an announcement will be played in the Proximus IMS network. To end the call 2 scenarios are possible:

- During the announcement the calling user ends the call by sending a *Cancel* message.
- At the end of the announcement the Proximus IMS network tears down the call by sending a *404 Not found* response.

Remark: According to ETSI TS 183 011 [37] the response in case of OCB should be *603 Decline*. In future evolution of the Proximus IMS network the usage of *603 Decline* instead of *404 Not found* is considered.

##### 6.4.4.1. OCB with call teardown during announcement

Figure 9 shows the expected message flow for an OCB with call teardown during announcement.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.



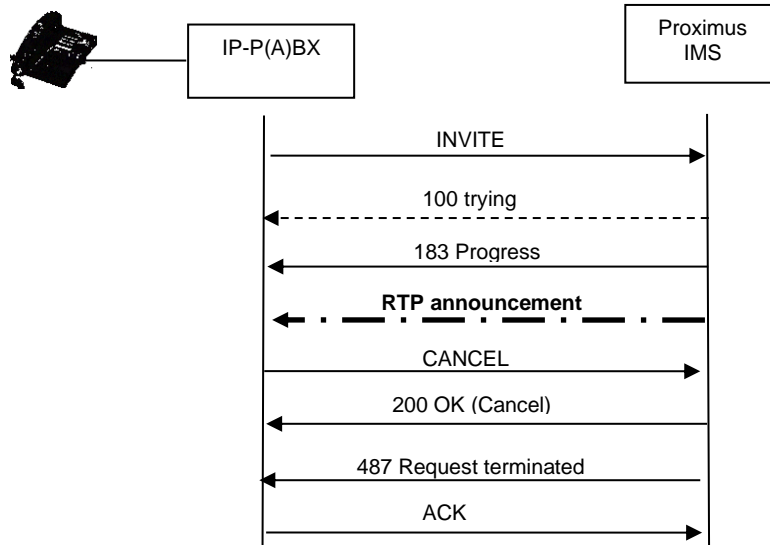


Figure 9: Basic call/session set-up to barred destination

In order to set-up a call the IP-P(A)BX sends an *INVITE* message to the Proximus IMS network. The Proximus IMS network returns a *100 Trying* response.

#### Live example

Request-Line: *INVITE sip:+3123456789@ims.belgacom.be SIP/2.0*

#### Message Header

Via: *SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-113020-2084*

From: "Dory" <*sip:+3225016490@dory.gent.be*>;tag=113020-59

To: <*sip:+3123456789@ims.belgacom.be*>

Call-ID: *113020-1104*

CSeq: *102 INVITE*

User-Agent: *Dory*

Contact: <*sip:+3225016490@10.127.32.100:5061*>

Expires: *15*

Allow: *CANCEL,BYE,INVITE, ACK*

Content-Type: *application/sdp*

Content-Length: *193*

Accept: *application/sdp*

#### Message Body

#### Session Description Protocol

Session Description Protocol Version (v): *0*

Owner/Creator, Session Id (o): *Dory 27106 3566 IN IP4 10.127.32.100*

Session Name (s): *SIP Call*

Connection Information (c): *IN IP4 10.127.32.100*

Time Description, active time (t): *0 0*

Media Description, name and address (m): *audio 10010 RTP/AVP 8 101*

Media Attribute (a): *rtpmap:8 PCMA/8000*

Media Attribute (a): *rtpmap:101 telephone-event/8000*

Media Attribute (a): *fntp:101 0-15*

Media Attribute (a): *sendrecv*

Status-Line: *SIP/2.0 100 Trying*

#### Message Header

Via: *SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-113020-2084*

From: <*sip:+3225016490@dory.gent.be;user=phone*>;tag=113020-59

To: <sip:+3123456789@ims.belgacom.be>  
Call-ID: 113020-1104  
CSeq: 102 INVITE  
Content-Length: 0

When the Proximus IMS network invokes the OCB feature it returns a *183 Progress* response to the IP-P(A)BX and an announcement is played.

#### Live example

Status-Line: SIP/2.0 183 Session Progress

##### Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-113020-2084  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=113020-59  
To: <sip:+3123456789@ims.belgacom.be>;tag=SDk4itb99-1336732429-1536571816009  
Call-ID: 113020-1104  
CSeq: 102 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Content-Type: application/sdp  
Content-Length: 205  
Server: Alcatel-Lucent-HPSS/3.0.3  
Session: Media

##### Message Body

##### Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169323795 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13510 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a):ptime:20

During the announcement the calling user ends the call and the IP-P(A)BX sends a *CANCEL* message to the Proximus IMS network.

The Proximus IMS network returns a *200 OK* response in order to properly close the *CANCEL* transaction.

#### Live example

Request-Line: CANCEL sip:+3123456789@ims.belgacom.be SIP/2.0

##### Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-113020-2084  
From: "Dory" <sip:+3225016490@dory.gent.be>;tag=113020-59  
To: <sip:+3123456789@ims.belgacom.be>  
Call-ID: 113020-1104  
CSeq: 102 CANCEL  
User-Agent: Dory  
Content-Length: 0

Status-Line: SIP/2.0 200 OK

##### Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-113020-2084

From: "Dory" <sip:+3225016490@dory.gent.be>;tag=113020-59  
To: <sip:+3123456789@ims.belgacom.be>;tag=SDk4itb99-1336732429-1536571816009  
Call-ID: 113020-1104  
CSeq: 102 CANCEL  
Content-Length: 0

The Proximus IMS network sends a *487 Request terminated* response to the IP-P(A)BX in order to indicate that the *INVITE* transaction is terminated.

Live example

Status-Line: SIP/2.0 487 Request terminated  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-113020-2084  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=113020-59  
To: <sip:+3123456789@ims.belgacom.be>;tag=SDk4itb99-1336732429-1536571816009  
Call-ID: 113020-1104  
CSeq: 102 INVITE  
Content-Length: 0  
Server: Alcatel-Lucent-HPSS/3.0.3

The IP-P(A)BX sends an *ACK* message to the Proximus IMS network in order to properly close the *INVITE* transaction.

Live example

Request-Line: ACK sip:81.247.255.118:5060;transport=udp SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-113020-2084  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=113020-59  
To: <sip:+3123456789@ims.belgacom.be>;tag=SDk4itb99-1336732429-1536571816009  
Call-ID: 113020-1104  
CSeq: 102 ACK  
Content-Length: 0

### 6.4.4.2. OCB with call teardown at end of announcement

Figure 10 shows the expected message flow for an OCB with call teardown at end of announcement.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

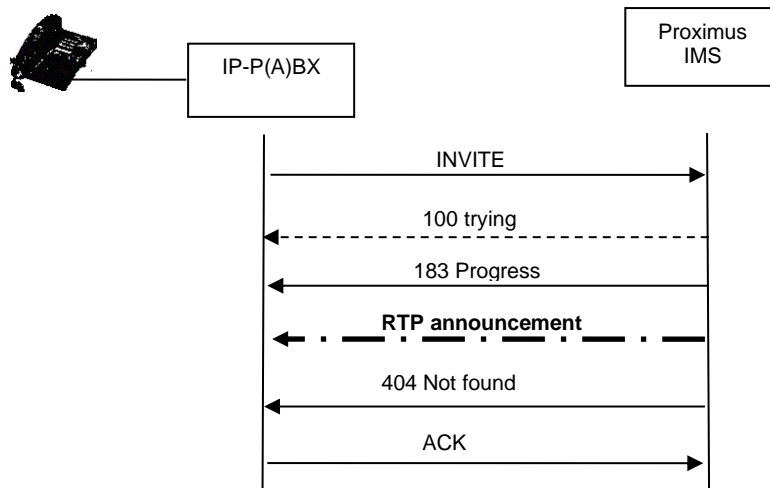


Figure 10: Basic call/session set-up to barred destination

In order to set-up a call the IP-P(A)BX sends an *INVITE* message to the Proximus IMS network. The Proximus IMS network returns a *100 Trying* response.

#### Live example

##### Session Initiation Protocol (INVITE)

Request-Line: INVITE sip:+3123456789@ims.belgacom.be SIP/2.0

##### Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-111424-2440

From: "Dory" <sip:+3225016490@dory.gent.be>;tag=111424-64

To: <sip:+3123456789@ims.belgacom.be>

Call-ID: 111424-5507

CSeq: 102 INVITE

User-Agent: Dory

Contact: <sip:+3225016490@10.127.32.100:5061>

Expires: 15

Allow: CANCEL,BYE,INVITE, ACK

Content-Type: application/sdp

Content-Length: 193

Accept: application/sdp

##### Message Body

##### Session Description Protocol

Session Description Protocol Version (v): 0

Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100

Session Name (s): SIP Call

Connection Information (c): IN IP4 10.127.32.100

Time Description, active time (t): 0 0

Media Description, name and address (m): audio 10010 RTP/AVP 8 101

Media Attribute (a): rtpmap:8 PCMA/8000

Media Attribute (a): rtpmap:101 telephone-event/8000

Media Attribute (a): fmp:101 0-15  
Media Attribute (a): sendrecv

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-111424-2440  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=111424-64  
To: <sip:+3123456789@ims.belgacom.be>  
Call-ID: 111424-5507  
CSeq: 102 INVITE  
Content-Length: 0

When the Proximus IMS network invokes the OCB feature it returns a *183 Progress* response to the IP-P(A)BX and an announcement is played.

[Live example](#)

Status-Line: SIP/2.0 183 Session Progress

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-111424-2440  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=111424-64  
To: <sip:+3123456789@ims.belgacom.be>;tag=SDqsj5c99-1384087817-1536570860724  
Call-ID: 111424-5507  
CSeq: 102 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Content-Type: application/sdp  
Content-Length: 205  
Server: Alcatel-Lucent-HPSS/3.0.3  
Session: Media

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169275026 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13498 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): ptime:20

After the announcement the Proximus IMS network ends the call and sends a *404 Not Found* response to the IP-P(A)BX.

[Live example](#)

Status-Line: SIP/2.0 404 Not Found

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-111424-2440  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=111424-64  
To: <sip:+3123456789@ims.belgacom.be>;tag=SDqsj5c99-1384087817-1536570860724  
Call-ID: 111424-5507  
CSeq: 102 INVITE  
Content-Length: 0  
Server: Alcatel-Lucent-HPSS/3.0.3

The IP-P(A)BX sends an ACK message to the Proximus IMS network in order to properly close the INVITE transaction.

#### Live example

```
Request-Line: ACK sip:81.247.255.118:5060;transport=udp SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-111424-2440
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=111424-64
To: <sip:+3123456789@ims.belgacom.be>;tag=SDqsj5c99-1384087817-1536570860724
Call-ID: 111424-5507
CSeq: 102 ACK
Content-Length: 0
```

## 6.4.5. Call Hold

The call hold feature shall be managed and handled by the IP-P(A)BX.

An IP-P(A)BX user may place an active call on hold and may retrieve a held call. The IP-P(A)BX shall provide “music on hold” or play an announcement for the held user. For these purposes, the Proximus IMS supports the use of re-INVITE transactions to modify the media description parameters for a call according to the held/retrieved state.

An IP-P(A)BX user involved in an active call may be placed on hold and may be retrieved. Depending on the holding user (external to the IP-P(A)BX) either the Proximus IMS or the holding user’s equipment is responsible to provide “music on hold” or to play an announcement for the held IP-P(A)BX user. For these purposes, the Proximus IMS supports the use of re-INVITE transactions to modify the media description parameters for a call according to the held/retrieved state.

### 6.4.5.1. Call Hold initiated by IP-P(A)BX user

An established public on-net or public off-net call can be put on Hold by the IP-P(A)BX user. The user which is put on hold will receive music or an announcement, provided by the IP-P(A)BX. There are two methods of providing music/announcements on hold:

- the IP-P(A)BX may connect the user on hold to its media resource (the music/announcement generator) using the re-INVITE mechanism to change the media description parameters:
  - announce a different IP address (used for sourcing the RTP stream containing the music/announcement)
  - add an *a=sendonly* attribute to the media description in order to stop receiving RTP from the external user.

In this case resuming the call is accomplished using the re-INVITE mechanism again in order to restore the original media description parameters:

- use the original IP address for the RTP stream
  - add an *a=sendrecv* attribute to the media description in order to resume both way RTP traffic.
- the IP-P(A)BX may connect the user on hold to its media resource (the music/announcement generator) without further signalling between the IP-P(A)BX and the Proximus IMS network. Resuming the call is accomplished also without further signalling.

In case the media stream (RTP) for music/announcement is sourced by the IP-P(A)BX from a different IP address then the media stream for voice the usage of the re-INVITE mechanism is **mandatory** in order to avoid problems with the security features (e.g. pinholing) performed at the edge of the Proximus IMS network.

In all other cases the usage of the re-INVITE mechanism is **strongly recommended** in order to avoid useless RTP streams.

Note: In the context of the Call Hold service putting a line on hold can also be accomplished by sending a re-INVITE containing in the SDP information 0.0.0.0 as IP address. The 200 OK response should contain in the SDP information an *a=inactive* attribute as a result. As this is an old mechanism it is **strongly discouraged** and support of this mechanism is being discontinued.

Figure 11 shows the expected message flow for a hold/retrieve performed by an IP-P(A)BX user.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

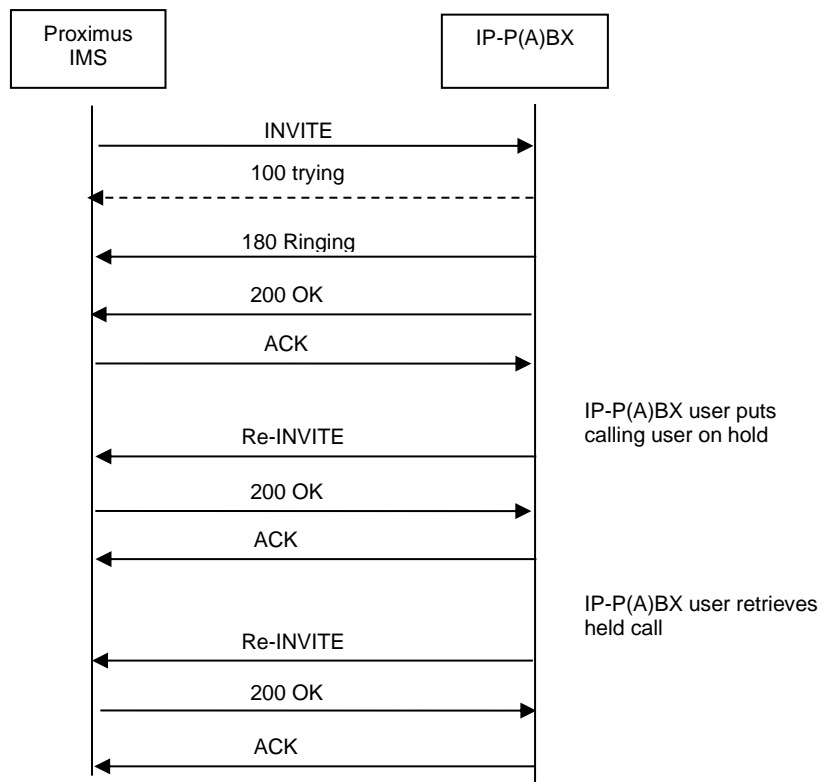


Figure 11: Hold/retrieve by IP-P(A)BX user example

An incoming call to an IP-P(A)BX user is set-up. When answering the call the 200 OK response contains in the SDP part the IP address of the called IP-P(A)BX user used for the media stream:

- *Connection Information* = IN IP4 <IP address of called user>

Live example

```

Request-Line: INVITE sip:+3225016490@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKbcjq1oe7ra83s6mj53qvnshcl7
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-
To: "3225016490 ." <sip:+3225016490@ims.belgacom.be;user=phone>
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043
CSeq: 565942034 INVITE
Contact: <sip:81.247.255.118:5060;transport=udp>
    
```

Privacy: none  
Supported: 100rel  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 294  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169727528 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13556 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKbcjq10e7ra83s6mj53qvnshcl7  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-  
1536582692387-  
To: "3225016490 ." <sip:+3225016490@ims.belgacom.be;user=phone>  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942034 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 180 Ringing

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKbcjq10e7ra83s6mj53qvnshcl7  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-  
1536582692387-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942034 INVITE  
Contact: <sip:+32475339778@10.127.32.100:5061>  
Content-Length: 0

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKbcjq10e7ra83s6mj53qvnshcl7  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-  
1536582692387-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942034 INVITE  
User-Agent: Dory



Contact: <sip:+32475339778@10.127.32.100:5061>  
Content-Type: application/sdp  
Content-Length: 193  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10140 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmtp:101 0-15  
Media Attribute (a): sendrecv

Request-Line: ACK sip:+32475339778@10.127.32.100:5061;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bK92n7sv6038mfid7j2fa3a36mv6-1pef99g5  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942034 ACK  
Contact: <sip:81.247.255.118:5060>  
Max-Forwards: 68  
Content-Length: 0  
User-Agent: Alcatel-Lucent-HPSS v3.0.3

When putting the call on hold the IP-P(A)BX sends a (re)INVITE message to the Proximus IMS network in order to change the session description parameters. The connection information is changed to contain the IP address of the media resource of the IP-P(A)BX that will be sourcing the “music on hold” and the media attribute is set to *sendonly*.

- *Connection Information* = IN IP4 <IP address of the media resource of the IP-P(A)BX>
- *Media Attribute* = *sendonly*

Note: changing the media attribute from *sendrecv* to *sendonly* is not mandatory but recommended in order to stop an unnecessary media stream.

#### Live example

Request-Line: INVITE sip:+32475339778@ims.belgacom.be SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-143144-8060  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12  
To: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942035 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Expires: 15  
Allow: CANCEL,BYE,INVITE, ACK  
Content-Type: application/sdp

Content-Length: 205  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10010 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): **sendonly**  
Data: odoa

The Proximus IMS network returns a 200 OK response acknowledging the changed media description parameters: The media attribute is set to *recvonly*.

- *Media Attribute* = *recvonly*

#### Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-143144-8060  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12  
To: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942035 INVITE  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept:  
application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,multipart/mixed  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Content-Type: application/sdp  
Content-Length: 259  
Server: Alcatel-Lucent-HPSS/3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169727528 2 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13556 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): **recvonly**  
Media Attribute (a): ptime:20

The IP-P(A)BX sends an ACK message to the Proximus IMS network in order to properly close the *INVITE* transaction.

### Live example

```
Request-Line: ACK sip:81.247.255.118:5060;transport=udp SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-143144-4598
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12
To: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043
CSeq: 565942035 ACK
Content-Length: 0
```

When retrieving the held call the IP-P(A)BX sends a (re)INVITE message to the Proximus IMS network in order to change the session description parameters. The connection information is changed back to contain the IP address of the called user and the media attribute is set to *sendrecv*.

- *Connection Information* = IN IP4 <IP address of called user>
- *Media Attribute* = *sendrecv*

### Live example

```
Request-Line: INVITE sip:+32475339778@ims.belgacom.be SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-143148-7188
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12
To: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043
CSeq: 565942036 INVITE
User-Agent: Dory
Contact: <sip:+3225016490@10.127.32.100:5061>
Expires: 15
Allow: CANCEL,BYE,INVITE, ACK
Content-Type: application/sdp
Content-Length: 205
Accept: application/sdp
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100
Session Name (s): SIP Call
Connection Information (c): IN IP4 10.127.32.100
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 10010 RTP/AVP 8 101
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15
Media Attribute (a): sendrecv
```

The Proximus IMS network returns a *200 OK* response acknowledging the changed media description parameters: The media attribute is set to *sendrecv*.

- *Media Attribute* = *sendrecv*

### Live example

```
Status-Line: SIP/2.0 200 OK
Message Header
```

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-143148-7188  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12  
To: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942036 INVITE  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,multipart/mixed  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Content-Type: application/sdp  
Content-Length: 259  
Server: Alcatel-Lucent-HPSS/3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169727528 3 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13556 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): **sendrecv**  
Media Attribute (a): ptime:20

The IP-P(A)BX sends an ACK message to the Proximus IMS network in order to properly close the *INVITE* transaction.

#### Live example

Request-Line: ACK sip:81.247.255.118:5060;transport=udp SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-143149-9528  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=143137-12  
To: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SDhcgve01-1018503028-1536582692387-  
Call-ID: SDhcgve01-42b247ca3ae77372a77b2d9abaded76a-aobq043  
CSeq: 565942036 ACK  
Content-Length: 0

### 6.4.5.2. Call Hold initiated by a user external to the IP-P(A)BX

Figure 12 shows the expected message flow for a hold/retrieve performed by an external user.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

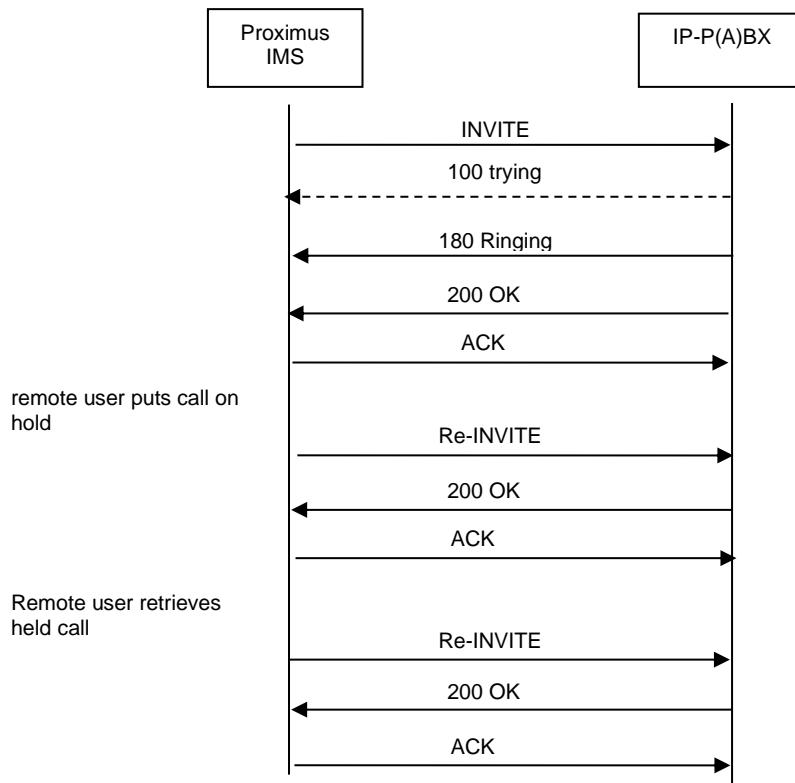


Figure 12: Hold/retrieve by remote user

An incoming call to an IP-P(A)BX user is set-up. When answering the call the 200 OK response contains in the SDP part the IP address of the called IP-P(A)BX user used for the media stream:

- *Connection Information* = IN IP4 <IP address of called user>

Live example

```

Request-Line: INVITE sip:+3225016490@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKv5tsaapc45fprseqg62fkng1d1
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-
To: "3225016490 ."<sip:+3225016490@ims.belgacom.be;user=phone>
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043
CSeq: 567247437 INVITE
Contact: <sip:81.247.255.118:5060;transport=udp>
Privacy: none
Supported: 100rel
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
    
```

Accept:  
application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,multipart/mixed

Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 294  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169849703 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13576 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKv5tsaapc45fprseqg62fkng1d1  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: "3225016490 ." <sip:+3225016490@ims.belgacom.be;user=phone>  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247437 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 180 Ringing

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKv5tsaapc45fprseqg62fkng1d1  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247437 INVITE  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Content-Length: 0

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKv5tsaapc45fprseqg62fkng1d1  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247437 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Content-Type: application/sdp  
Content-Length: 193

Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10000 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): sendrecv

Request-Line: ACK sip:+3225016490@10.127.32.100:5061;user=phone SIP/2.0

Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKcre6lu619m793mamd7dg9bot33-1pj506h5  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247437 ACK  
Contact: <sip:81.247.255.118:5060>  
Max-Forwards: 68  
Content-Length: 0  
User-Agent: Alcatel-Lucent-HPSS v3.0.3

When the call is put on hold by the remote user the IP-P(A)BX receives a (re)INVITE message from the Proximus IMS network in order to change the session description parameters. The media attribute is set to *sendonly*.

- *Media Attribute = sendonly*

Note: The IP-P(A)BX user will receive “music on hold” provided by the media resource function of the Proximus IMS network. Connection to this media resource is invisible for the IP-P(A)BX and the media stream IP address used on the UNI interface remains the same. Therefore, the *Connection Information* is not changed.

#### Live example

Request-Line: INVITE sip:+3225016490@10.127.32.100:5061;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKcre6lu619m793mamd7dg9bot33-1pj587d5  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247438 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 306  
User-Agent: Alcatel-Lucent-HPSS v3.0.3

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169849703 2 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13576 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): **sendonly**  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

The IP-P(A)BX shall return a 200 OK response to the Proximus IMS network in order to acknowledge the changed media description parameter and shall suspend sending RTP packets.

- Media Attribute = recvnly

Live example

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKcre6lu619m793mamd7dg9bot33-1pj587d5  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247438 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Content-Type: application/sdp  
Content-Length: 205  
Accept: application/sdp

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10000 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): **recvnly**

The Proximus IMS network sends an ACK message to the IP-P(A)BX in order to properly close the INVITE transaction.

Live example

Request-Line: ACK sip:+3225016490@10.127.32.100:5061;user=phone SIP/2.0



Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKcre6lu619m793mamd7dg9bot33-1pj587h5  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247438 ACK  
Contact: <sip:81.247.255.118:5060>  
Max-Forwards: 68  
Content-Length: 0  
User-Agent: Alcatel-Lucent-HPSS v3.0.3

When the call is retrieved by the remote user the IP-P(A)BX receives a (re)INVITE message from the Proximus IMS network in order to change the session description parameters. The media attribute is set to *sendrecv*.

- *Media Attribute* = *sendrecv*

Note: The IP-P(A)BX user will be reconnected to the remote user by the Proximus IMS network. This is invisible for the IP-P(A)BX and the media stream IP address used on the UNI interface remains the same. Therefore, the *Connection Information* is not changed.

#### Live example

Request-Line: INVITE sip:+3225016490@10.127.32.100:5061;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKcre6lu619m793mamd7dg9bot33-1pj507d5  
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11  
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043  
CSeq: 567247439 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 306  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 169849703 3 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13576 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): **sendrecv**  
Media Attribute (a): maxptime:30

Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

The IP-P(A)BX shall return a 200 OK response to the Proximus IMS network in order to acknowledge the changed media description parameter and shall resume sending RTP packets.

- *Media Attribute* = sendrecv

#### Live example

```
Status-Line: SIP/2.0 200 OK
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKcre6lu619m793mamd7dg9bot33-1pj507d5
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043
CSeq: 567247439 INVITE
User-Agent: Dory
Contact: <sip:+3225016490@10.127.32.100:5061>
Content-Type: application/sdp
Content-Length: 205
Accept: application/sdp
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100
Session Name (s): SIP Call
Connection Information (c): IN IP4 10.127.32.100
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 10000 RTP/AVP 8 101
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15
Media Attribute (a): sendrecv
```

The Proximus IMS network sends an ACK message to the IP-P(A)BX in order to properly close the *INVITE* transaction.

#### Live example

```
Request-Line: ACK sip:+3225016490@10.127.32.100:5061;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKcre6lu619m793mamd7dg9bot33-1pj507h5
From: <sip:+32475339778@fed.ipcx.be:51000;user=phone>;tag=SD6di3a01-785921605-1536585303192-
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=151508-11
Call-ID: SD6di3a01-6db997da2b5a2a7fc3dd3d4bee4125a5-aobq043
CSeq: 567247439 ACK
Contact: <sip:81.247.255.118:5060>
Max-Forwards: 68
Content-Length: 0
User-Agent: Alcatel-Lucent-HPSS v3.0.3
```

## 6.4.6. Conference call

The conference feature shall be managed and handled by the IP-P(A)BX. In case external users are involved in a conference call, the incoming and/or outgoing calls to/from the IP-P(A)BX will be treated like normal basic calls by the Proximus IMS.

## 6.4.7. Call transfer

The call transfer feature shall be managed and handled by the IP-P(A)BX.

The following behaviour is valid for all flavours (e.g. with or without consultation) of call transfer executed by the IP-P(A)BX.

In case a call involving an IP-P(A)BX user is transferred (by that IP-P(A)BX user) to a destination **external** to the IP-P(A)BX, the IP-P(A)BX shall:

- Either set-up a new call to the transferred-to destination, and:
  - o either use a re-INVITE to transfer the RTP stream of the original call to the new destination. No *Diversion* header nor *History-info* header shall be included in the outgoing call. The calling line identity possibly shown to the transferred-to destination will be the identity of the forwarding IP-P(A)BX user (or the IP-P(A)BX general number).
  - o or not use a re-INVITE and handle the transfer of the RTP streams autonomously.
- or use a REFER message containing the Referred-by header and Referred-to header in order to allow the Proximus IMS network to redirect the call to the transferred-to-number. The calling line identity possibly shown to the transferred-to destination will be the identity of the original calling user.

### 6.4.7.1. Call transfer without using re-INVITE

Figure 13 shows the expected message flow for a call transfer without using re-INVITE.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

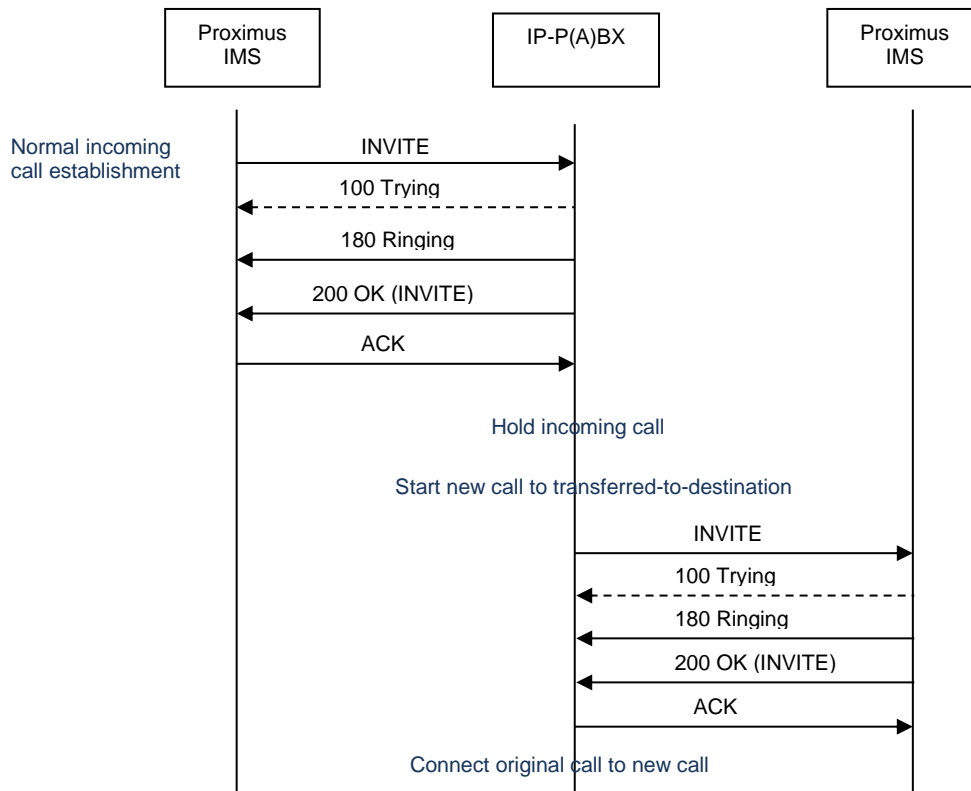


Figure 13: Call transfer without using re-INVITE

In this call transfer scenario, the transfer is in fact completely invisible for the Proximus IMS network. The IP-P(A)BX acts on 2 independent calls and remains in the call path, for signaling as well as for the RTP streams even after the call transfer has been completed. Consequently this scenario is not optimal from a bandwidth usage point of view.

### 6.4.7.2. Call transfer using re-INVITE

Figure 14 shows the expected message flow for a call transfer using re-INVITE.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

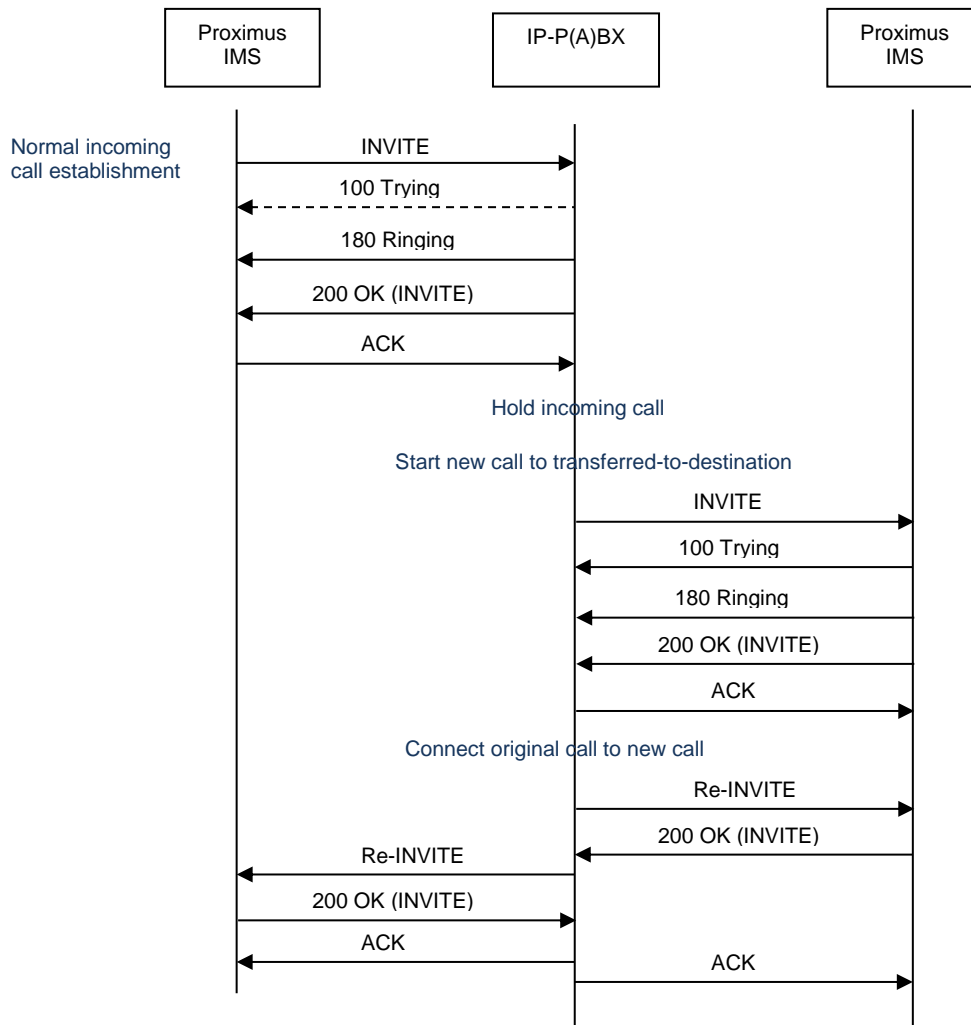


Figure 14: Call transfer using re-INVITE

To transfer an incoming established call the IP-P(A)BX first puts the established call on hold and then sends an *INVITE* message to the Proximus IMS network in order to start a new call to the transferred-to destination.

The initial call is set-up according to § 6.3.3 or § 6.3.4 above. The SDP information in the *INVITE* message contains the IP address and port number on which the Proximus IMS network (i.e. the SBC) wants to receive the RTP stream. The SDP information in the *200 OK* response contains the IP address and port number on which the IP-P(A)BX user wants to receive the RTP stream

Live example

```
Request-Line: INVITE sip:+3225016490@ims.belgacom.be;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKggjmccbaq5dl8qi9td42pdhaki
From: "3292212981." <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-
To: "3225016490." <sip:+3225016490@ims.belgacom.be;user=phone>
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043
CSeq: 650668686 INVITE
Contact: <sip:81.247.255.118:5060;transport=udp>
Privacy: none
```

Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 252  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 172753593 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13660 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a):ptime:20

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKggjmccbaq5dl8qi9td42pdhak1  
From: "3292212981." <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-  
1258464469-1536752145690-  
To: "3225016490." <sip:+3225016490@ims.belgacom.be;user=phone>  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668686 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 180 Ringing

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKggjmccbaq5dl8qi9td42pdhak1  
From: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-  
1536752145690-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668686 INVITE  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Content-Length: 0

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKggjmccbaq5dl8qi9td42pdhak1  
From: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-  
1536752145690-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668686 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Content-Type: application/sdp  
Content-Length: 193  
Accept: application/sdp  
Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10000 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): sendrecv

Request-Line: ACK sip:+3225016490@10.127.32.100:5061;user=phone SIP/2.0

Message Header

Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKupv8buet2mqS7va8b8qjrov2a3-6iil67i5  
From: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668686 ACK  
Contact: <sip:81.247.255.118:5060>  
Max-Forwards: 68  
Content-Length: 0  
User-Agent: Alcatel-Lucent-HPSS v3.0.3

When putting the initial call on hold, the IP-P(A)BX uses a *re-INVITE* message in order to connect the calling user to its media resource responsible for providing “music on hold”. The *re-INVITE* message contains:

- *Connection information* = the IP address of the media resource of the IP-P(A)BX
- *Media port* = the port number of the media resource of the IP-P(A)BX
- *Media Attribute* = sendonly

Live example

Request-Line: INVITE sip:+3225016490@ims.belgacom.be SIP/2.0

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133553-5152  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
To: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668687 INVITE  
User-Agent: Dory  
Contact: <sip:+3292212981@10.127.32.100:5061>  
Expires: 15  
Allow: CANCEL,BYE,INVITE, ACK  
Content-Type: application/sdp  
Content-Length: 205  
Accept: application/sdp

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0

Media Description, name and address (m): audio 10010 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): **sendonly**

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133553-5152  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
To: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668687 INVITE  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Content-Type: application/sdp  
Content-Length: 217  
Server: Alcatel-Lucent-HPSS/3.0.3

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 172753593 2 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13660 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): **recvonly**  
Media Attribute (a):ptime:20

Request-Line: ACK sip:81.247.255.118:5060;transport=udp SIP/2.0

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133554-1385  
From: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
To: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668687 ACK  
Content-Length: 0

The new call to the transferred-to destination is set-up according to § 6.3.3 or § 6.3.4 above. The SDP information in the *INVITE* message contains the IP address and port number on which the IP-P(A)BX user wants to receive the RTP stream. The SDP information in the *200 OK* response contains the IP address and port number on which the Proximus IMS network (i.e. the SBC) wants to receive the RTP stream.

Live example

Request-Line: INVITE sip:+32475339778@ims.belgacom.be SIP/2.0

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133554-2727  
From: "Dory" <sip:+3225016490@dory.gent.be>;tag=133554-35  
To: <sip:+32475339778@ims.belgacom.be>  
Call-ID: 133554-760



CSeq: 102 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Expires: 15  
Allow: CANCEL,BYE,INVITE, ACK  
Content-Type: application/sdp  
Content-Length: 193  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Dory 27106 3566 IN IP4 10.127.32.100  
Session Name (s): SIP Call  
Connection Information (c): IN IP4 10.127.32.100  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 10010 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133554-2727  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=133554-35  
To: <sip:+32475339778@ims.belgacom.be>  
Call-ID: 133554-760  
CSeq: 102 INVITE  
Content-Length: 0

Status-Line: SIP/2.0 183 Session Progress

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133554-2727  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=133554-35  
To: <sip:+32475339778@ims.belgacom.be>;tag=SDts77c99-786978462-1536752157082  
Call-ID: 133554-760  
CSeq: 102 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Content-Type: application/sdp  
Content-Length: 247  
Server: Alcatel-Lucent-HPSS/3.0.3  
Session: Media

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 172753964 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13662 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133554-2727  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=133554-35  
To: <sip:+32475339778@ims.belgacom.be>;tag=SDts77c99-786978462-1536752157082  
Call-ID: 133554-760  
CSeq: 102 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml  
Content-Type: application/sdp  
Content-Length: 247  
Server: Alcatel-Lucent-HPSS/3.0.3

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 172753964 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13662 RTP/AVP 8 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - -  
Media Attribute (a): ptime:20

Request-Line: ACK sip:81.247.255.118:5060;transport=udp SIP/2.0

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133559-6866  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=133554-35  
To: <sip:+32475339778@ims.belgacom.be>;tag=SDts77c99-786978462-1536752157082  
Call-ID: 133554-760  
CSeq: 102 ACK  
Content-Length: 0

To perform the call transfer, the IP-P(A)BX sends an *INVITE* message without SDP information to the transferred-to destination, in order to collect the SDP information.

Live example

Request-Line: INVITE sip:+32475339778@ims.belgacom.be SIP/2.0

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133559-9064  
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=133554-35  
To: <sip:+32475339778@ims.belgacom.be>;tag=SDts77c99-786978462-1536752157082  
Call-ID: 133554-760  
CSeq: 103 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Allow: CANCEL,BYE,INVITE,ACK  
Expires: 15  
Content-Length: 0  
Accept: application/sdp

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133559-9064

From: <sip:+3225016490@dory.gent.be;user=phone>;tag=133554-35  
To: <sip:+32475339778@ims.belgacom.be>;tag=SDts77c99-786978462-1536752157082  
Call-ID: 133554-760  
CSeq: 103 INVITE  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Content-Type: application/sdp  
Content-Length: 306  
Server: Alcatel-Lucent-HPSS/3.0.3  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 172753964 2 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13662 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmp:101 0-15  
Media Attribute (a): **sendrecv**  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

The IP-P(A)BX sends an *INVITE* message to the original calling user, including the collected SDP information.

- *Connection information* = the IP address of the media resource of the Proximus IMS network
- *Media port* = the port number of the media resource of the Proximus IMS network
- *Media Attribute* = sendrecv

#### Live example

Request-Line: INVITE sip:+3225016490@ims.belgacom.be;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKggjmccbaq5dl8qi9td42pdhak1  
From: "3292212981." <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-  
To: "3225016490." <sip:+3225016490@ims.belgacom.be;user=phone>  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobqo43  
CSeq: 650668688 INVITE  
Contact: <sip:81.247.255.118:5060;transport=udp>  
Privacy: none  
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE  
Accept: application/media\_control+xml,application/sdp,application/vnd.etsi.pstn+xml,  
multipart/mixed  
Max-Forwards: 68  
Content-Type: application/sdp  
Content-Length: 252  
User-Agent: Alcatel-Lucent-HPSS v3.0.3  
Recv-Info: x-broadworks-client-session-info  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0

Owner/Creator, Session Id (o): BroadWorks 172753964 2 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13662 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmtp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmtp:101 0-15  
Media Attribute (a): **sendrecv**  
Media Attribute (a): maxptime:30  
Media Attribute (a): silenceSupp:off - - - -  
Media Attribute (a): ptime:20

The Proximus IMS network returns a 200 OK response including SDP information.

Live example

Status-Line: SIP/2.0 200 OK  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKggjmccbaq5dl8qi9td42pdhak1  
From: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668688 INVITE  
User-Agent: Dory  
Contact: <sip:+3225016490@10.127.32.100:5061>  
Content-Type: application/sdp  
Content-Length: 193  
Accept: application/sdp  
Message Body  
Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): BroadWorks 172753593 1 IN IP4 81.247.255.118  
Session Name (s): -  
Connection Information (c): IN IP4 81.247.255.118  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 13660 RTP/AVP 8 18 101  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmtp:18 annexb=no  
Media Attribute (a): rtpmap:101 telephone-event/8000  
Media Attribute (a): fmtp:101 0-15  
Media Attribute (a): ptime:20  
  
Request-Line: ACK sip:+3225016490@10.127.32.100:5061;user=phone SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 81.247.255.118:5060;branch=z9hG4bKupv8buet2mq57va8b8qjrov2a3-61il67i5  
From: <sip:+3292212981@ies.belgacom.be;user=phone>;tag=SDhtg5b01-1258464469-1536752145690-  
To: <sip:+3225016490@ims.belgacom.be;user=phone>;tag=133546-15  
Call-ID: SDhtg5b01-d9dd1c27b5ac34086a6a133303ec8724-aobq043  
CSeq: 650668688 ACK  
Contact: <sip:81.247.255.118:5060>  
Max-Forwards: 68

### Content-Length: 0

The IP-P(A)BX uses the SDP information received from the Proximus IMS network in order to shortcut the 2 RTP streams.

- *Connection information* = the IP address of the media resource of the Proximus IMS network
- *Media port* = the port number of the media resource of the Proximus IMS network
- *Media Attribute* = sendrecv

### Live example

```
Request-Line: ACK sip:81.247.255.118:5060;transport=udp SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.32.100:5061;branch=z9hG4bK-133559-6866
From: <sip:+3225016490@dory.gent.be;user=phone>;tag=133554-35
To: <sip:+32475339778@ims.belgacom.be>;tag=SDts77c99-786978462-1536752157082
Call-ID: 133554-760
CSeq: 103 ACK
Content-Type: application/sdp
Content-Length: 193
Accept: application/sdp
Message Body
Session Description Protocol
  Session Description Protocol Version (v): 0
  Owner/Creator, Session Id (o): BroadWorks 172753593 1 IN IP4 81.247.255.118
  Session Name (s): -
  Connection Information (c): IN IP4 81.247.255.118
  Time Description, active time (t): 0 0
  Media Description, name and address (m): audio 13660 RTP/AVP 8 18 101
  Media Attribute (a): rtpmap:8 PCMA/8000
  Media Attribute (a): rtpmap:18 G729/8000
  Media Attribute (a): fmp:18 annexb=no
  Media Attribute (a): rtpmap:101 telephone-event/8000
  Media Attribute (a): fmp:101 0-15
  Media Attribute (a):ptime:20
```

### 6.4.7.3. Call transfer using REFER

Figure 15 shows the expected message flow for a call transfer using REFER.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

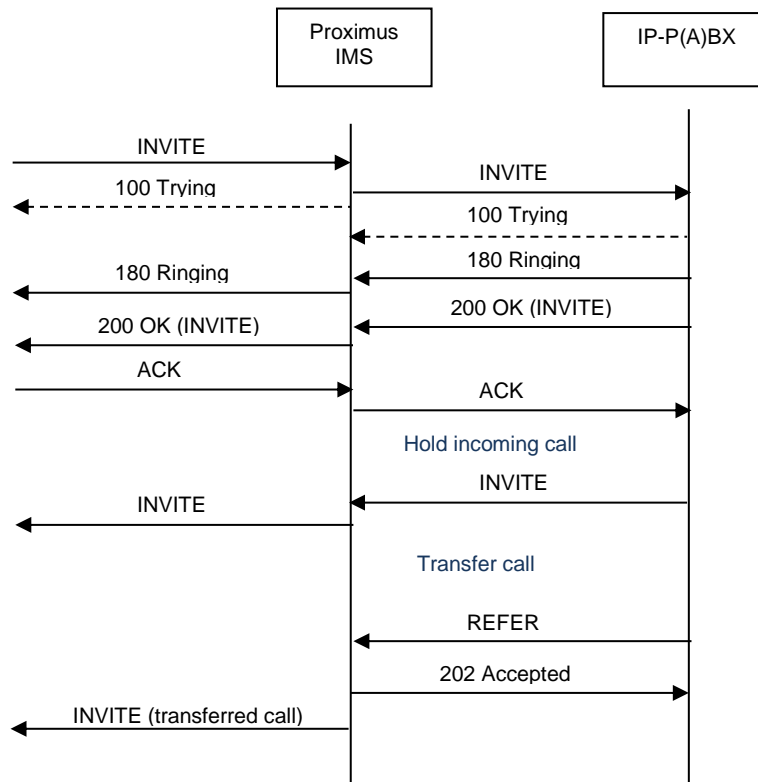


Figure 15: Call transfer using *REFER*

## 6.5. Call admission control

Call Admission Control concerns controlling the number of simultaneous calls that can be issued or received by an IP-P(A)BX. The maximum number of simultaneous calls can be set per IP-P(A)BX and per Trunk Group.

### 6.5.1. Incoming call to the IP-P(A)BX rejected by call admission control

In case an incoming call to an IP-P(A)BX exceeds the maximum number of simultaneous calls:

- either the call is rejected in the Proximus IMS network. The call will **not** be offered to the IP-P(A)BX. The calling party will receive a busy indication.
- or the “forced” rerouting feature forwards the call to a new destination, without offering the call to the IP-P(A)BX.

## 6.5.2. Outgoing call from the IP-P(A)BX rejected by call admission control

In case an outgoing call from an IP-P(A)BX is rejected in the Proximus IMS network by call admission control, the Proximus IMS network will return a *403 Forbidden* response to the IP-P(A)BX.

## 6.6. CLI screening

The identity of the calling user received from the IP-P(A)BX in the *From* header and *P-asserted-ID*, is verified by the Proximus IMS network by means of a screening function. This function consists of checking if the calling number belongs to the number range(s) defined for the IP-P(A)BX.

If the screening check gives a negative result, the network overwrites the calling identity in the *From* header and *P-asserted-ID* header with a default value for the IP-P(A)BX.

As some services at the destination side may depend on the calling identity it is important that the IP-P(A)BX sends the correct information in the *From* header and the *P-Asserted-ID* header.

## 6.7. FAX support

See “PXM IMS VoIP – UNI specification – Fax over IP” [36]

## 6.8. Emergency calls

Please refer to document “PXM IMS Corporate VOIP - UNI specification - General” [1] version 2.0 or later, chapter 14.

## 6.9. DTMF

For transport of DTMF the following capabilities exist:

- Usage of RFC 4733 (inband signalling in RTP by use of events)
- Usage of G.711 codec (inband signalling in RTP)

Both methods of transporting DTMF shall be supported.

**Evolution:** Usage of the SIP INFO message (outband signalling; the info follows the path of signalling) is currently under study and may be announced in a later version of this document

## 6.10. Redundancy procedures

### 6.10.1. Introduction

For redundancy reasons pairs of session border controllers will be installed as entry point to the IMS Core network. At present 1 pair is installed in Brussels at the locations Marais and Stro. These SBCs will be Acme Session Director 9200s and will be deployed as 2 units (1 Active and 1 Hot Standby) at each location.

The IP PBXs can be deployed according to different redundancy schemes, according to the high availability requirements set by the enterprise. The Proximus IMS Geographic Redundancy functionality is able to handle the following PBX connectivity & redundancy schemes:

- Centralised call handling (all telephony services are handled by 1 IP-PBX)
  - Single IP-PBX – Active/Active SBCs
- Distributed call handling (all telephone services are handled by multiple IP-PBXs in a multisite environment)
  - Active/Standby IP-PBX(s) – Active/Active SBCs
  - Active/Active IP-PBX(s) – Active/Active SBCs

#### 6.10.1.1. Single IP-PBX – Active/Active SBCs

This scheme also known as load sharing over both SBCs, will protect against SBC device & connectivity failures, but it does not protect against complete PBX failure.

The IP-PBX needs to be able to contact at least 2 SIP gateways (SBCs) using 2 SIP trunks. Under normal conditions the IP-PBX shall distribute equally the traffic over both SIP trunks/SBCs.

In case of failure of one SIP trunk i.e. failure of one SBC (or the connectivity to it), all traffic shall be sent to the other SIP trunk/SBC.

Note; In some case it can be allowed to use another distribution of traffic e.g. 75%-25% or even 100%-0% instead of 50%-50%. However, the requirement to support 2 SIP trunks simultaneously remains the same. It is obvious that in the 100%-0% case the SIP trunk with 0% of normal traffic will carry SIP OPTIONS messages only (see 6.10.2)

#### 6.10.1.2. Active/Standby IP-PBX – Active/Active SBCs

This configuration contains multiple IP-PBXs in active/standby redundancy (duplication). This scheme allows the switch over from one IP-PBX to another. One IP-PBX is active (and primary server) and the other server is constantly in watchdog mode on standby. If the primary server fails, the standby automatically takes over.

Both IP-PBXs need to be able to contact at least 2 SIP gateways (SBCs) using 2 SIP trunks, Under normal conditions the active IP-PBX shall distribute equally the traffic over both SIP trunks/SBCs.

In case of failure of one SIP trunk i.e. failure of one SBC (or the connectivity to it), all traffic shall be sent to the other SIP trunk/SBC.

In case of IP-PBX failure the standby IP-PBX will take over and shall distribute the traffic over both SIP trunks/SBCs

Note; In some case it can be allowed to use another distribution of traffic e.g. 75%-25% or even 100%-0% instead of 50%-50%. However, the requirement to support 2 SIP trunks simultaneously remains the same. It is obvious that in the 100%-0% case the SIP trunk with 0% of normal traffic will carry SIP OPTIONS messages only (see 6.10.2)



### 6.10.1.3. Active/Active IP-PBX – Active/Active SBCs

This configuration contains multiple IP-PBXs in active/active redundancy (duplication). They act as a single IP-PBX system and are all actively processing calls,

Both IP-PBXs need to be able to contact at least 2 SIP gateways (SBCs) using 2 SIP trunks, Under normal conditions both IP-PBXs shall distribute equally their traffic over both SIP trunks/SBCs.

In case of failure of one SIP trunk i.e. failure of one SBC (or the connectivity to it), all traffic shall be sent to the other SIP trunk/SBC.

In case of IP-PBX failure the other IP-PBX will take over and shall distribute the traffic over both SIP trunks/SBCs.

Note; In some case it can be allowed to use another distribution of traffic e.g. 75%-25% or even 100%-0% instead of 50%-50%. However, the requirement to support 2 SIP trunks simultaneously remains the same. It is obvious that in the 100%-0% case the SIP trunk with 0% of normal traffic will carry SIP OPTIONS messages only (see 6.10.2)

## 6.10.2. Availability monitoring

### 6.10.2.1. Failure detection by SBC

In order to detect a SIP trunk failure the SBC uses normal traffic and SIP OPTIONS messages.

As long as there is normal traffic on the SIP trunk no OPTIONS messages will be sent.

Under normal conditions i.e. the IP-PBX sends non 5XX responses to the SBC.

In case of failure the IP-PBX might send a 5XX response or no response at all. In the latter case the SBC applies the normal retransmission mechanism as described in IETF RFC 3261.

In case of a 5XX response the SBC will propagate the response into the IMS network allowing a re-attempt via a different route.

#### **WARNING!**

In case of a 503 response with Retry-After header the SBC will consider the SIP link with the IP-PBX as out-of-service for the duration of the Retry-After header.

**The SBC will not sent/receive any traffic over the SIP link during this out-of-service period causing a temporary total outage !**

In order to avoid long outages of the SIP link it is strongly advised not to use the Retry-After header in any 5XX response, or to use a duration of less then 5 seconds.

After 1 minute without normal traffic on the SIP trunk the SBC sends an OPTIONS message.

Under normal conditions i.e. the IP-PBX sends a SIP response to the OPTIONS message the OPTIONS message is repeated after 1 minute as long as normal traffic is absent.

In case the IP-PBX fails to send a SIP response the SBC applies the normal retransmission mechanism as described in IETF RFC 3261. If there is still no response at the end of the retransmission scheme, the SBC will consider the SIP trunk as "out of service"

#### 6.10.2.2. Recovery detection by SBC

In order to detect a SIP trunk recovery the SBC uses SIP OPTIONS messages. 2 minutes after the failing OPTIONS message the SBC will send a new OPTIONS message. If this is successful the SIP trunk is considered to be “in service”, else the SBC will repeat the OPTIONS message every 4 minutes.

#### 6.10.2.3. Failure detection by IP-PBX

In order to detect a SIP trunk failure the IP-PBX shall use SIP OPTIONS messages. As long as there is normal traffic on the SIP trunk there is no need to send OPTIONS messages. The first OPTIONS message shall be sent after 1 minute without normal traffic on the SIP trunk.

Under normal conditions i.e. the SBC sends a SIP response to the OPTIONS message the OPTIONS message is repeated after 1 minute as long as normal traffic is absent.

In case the SBC fails to send a SIP response the IP-PBX shall apply the normal retransmission mechanism as described in IETF RFC 3261. If there is still no response at the end of the retransmission scheme, the IP-PBX will consider the SIP trunk as “out of service”

In case the IP-PBX does send OPTIONS messages while there is normal traffic the interval shall be 2 minutes or higher in order not to overload the SBC.

#### 6.10.2.4. Recovery detection by IP-PBX

In order to detect a SIP trunk recovery the IP-PBX uses SIP OPTIONS messages. 2 minutes after the failing OPTIONS message the IP-PBX shall send a new OPTIONS message. If this is successful the SIP trunk is considered to be “in service”, else the IP-PBX will repeat the OPTIONS message every 4 minutes.